Avaya Solution & Interoperability Test Lab

Application Notes for Mediatrix 4104 with Avaya Communication Server 1000 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 6.0 and Mediatrix 4104. Mediatrix 4104 is a VOIP gateway that allows analog phone lines connected to Mediatrix 4104 to be able to register, as a SIP Client endpoint, with the Communication Server 1000. The Mediatrix 4104 allows user to re-use existing analog phones which can place and receive calls from the Communication Server 1000 Release 6.0 non-SIP and SIP Line clients. The compliance tests focused on basic telephone features including transfer, and conference.

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.

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1. Introduction

These Application Notes provide detail configurations of Avaya Communication Server 1000 SIP Line R6.0 (hereafter refer to as CS1000) and Mediatrix 4104 (here after refer to as M4140).. M4104 allows analog phones to be tested against the non-SIP and SIP clients of the CS1000 SIP line 6.0. All the applicable telephony feature test cases of release 6.0 SIP line were executed on the analog phones that connected to M 4104 port, where applicable, to ensure the interoperability with CS1000.

1.1. Interoperability Compliance Testing

The focus of this testing is to verify that the M4104 is able to interoperate with the CS1000 SIP line system. The following areas were tested:

- The M4104 gateway must be able to be installed in the same local VLAN network the CS1000 successfully.
- Registration of the M4104 to the CS1000 SIP Line Gateway.
- Calls establishment of analog phones from Mediatrix 4104 with Avaya SIP and non-SIP phones of the CS1000.
- Calls establishment between analog phones and PSTN phones.

- Telephony features: Basic calls, conference, transfer, DTMF (dual tone multi frequency) transmission, voicemail with Message Waiting Indication (MWI) notification, busy, hold, speed dial, group call pickup, call waiting, ring again busy/no answer, multiple appearances Directory Number.
- Codec negotiation – G711 and G729.

1.2. Support

For technical support on Mediatrix M4104 gateway, please contact Media5 Corporation technical support at website http://www.media5corp.com or telephone: 1-819-829-8749.

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the M4104.

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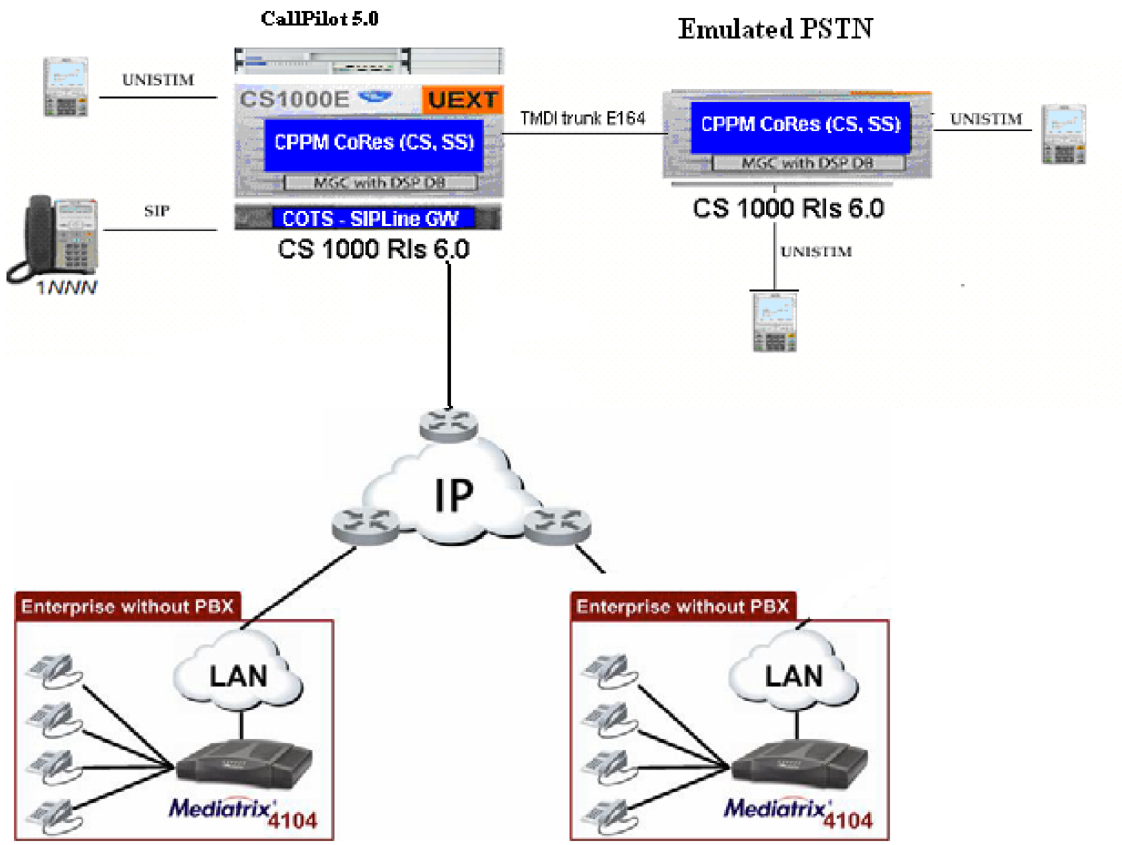

Figure 1

3. Equipment and Software Validated

System	Software Version
CS1000	• Call Server (CPPM): 6.00RJ • Signalling Server (CPPM): 6.00.18 • SIP Line Gateway (HP DL320)
Voicemail system	• CallPilot 5.0 system
11xx SIP client (Sigma) SIP soft-phones IP phones Analog phone	• 02.02.16.00 • SMC3456: v2.6 Build 53715 • 2050PC: 3.02.0045 • Northern Telecom, NTD 9519
Mediatrix 4104	• Dgw 2.0.9.144

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4. Configure Avaya CS 1000 - SIP LINE

This section describes the steps to configure SIP Line using CS 1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on CS 1000 system. For detailed information, see [1].

4.1. Prerequisite

- CS 1000 system has been upgraded to Release 6.0. For more information, see ***NN43041-458 Communication Server 1000E Software Upgrades.***
- A server which has been
  - Installed with CS 1000 Release 6.0 Linux Base.
  - Joined CS 1000 Release 6.0 Security Domain.
  - Deployed with SIP Line Application.

  For more information, see [6].
- Following packages are enabled in the keycode.
- If it has not been enable, please contact Avaya technical support at www.avaya.com

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

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

- Using internet browser, 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.
- Login with the username/password which was defined during the primary security server configuration. For more information, see [8].

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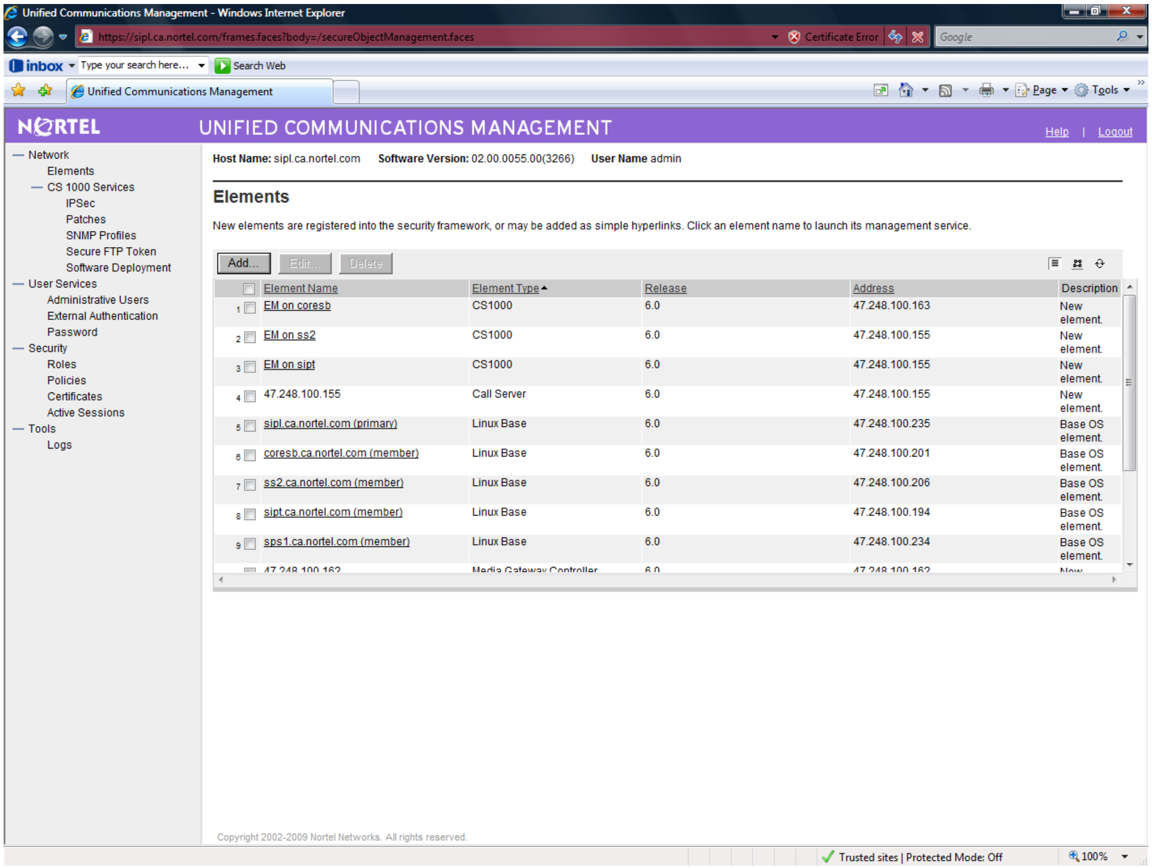

Figure 2: UCM Home Page

- On the Unified Communications Management page, under the Element Name column, click on the server name to navigate to Element Manager for that server. The CS 1000 Element Manager page appears as show in figure 3 below.

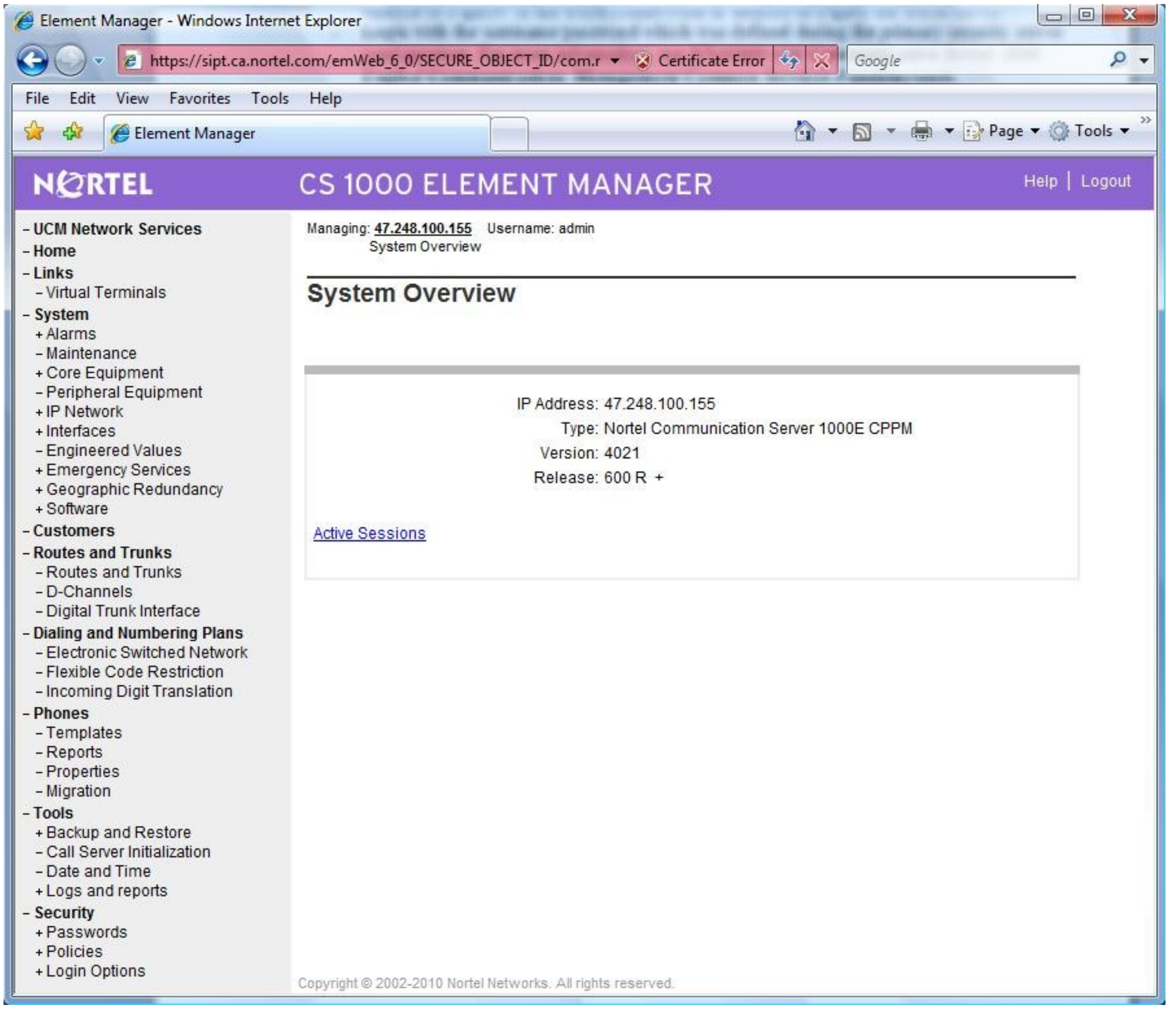

Figure 3: CS 1000 EM Home Page

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

- On the EM page, navigate to ***Customers*** 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 SIP Line **Root Domain** name in the ***Root Domain*** text box as shown in figure 4.

Figure 4: SIP Line Service in Customers Data Block

4.4. SIP Line Telephony Node Configuration

- On the EM page, navigate to ***System → IP Network → Nodes: Servers, Media Cards.***
- Click ***Add*** to add a new SIP Line Node to IP Telephony Nodes. To see the SIP Line node details, click on the SIP Line Node ID (not shown).
- Enter Node ID in the ***Node ID*** text box.
- Enter Call Server IP Address in the ***Call Server IP Address*** text box.
- Enter Node IP Address in the ***Node IP Address*** text box.
- Enter TLAN Subnet Mask in the ***Subnet Mask*** text box.
- Enter ELAN Gateway IP Address in the ***Gateway IP Address*** text box.
- Enter ELAN Subnet Mask in the ***Subnet Mask*** text box.
- Check **SIP Line** check box to enable SIP Line for this Node as show in figure 5.

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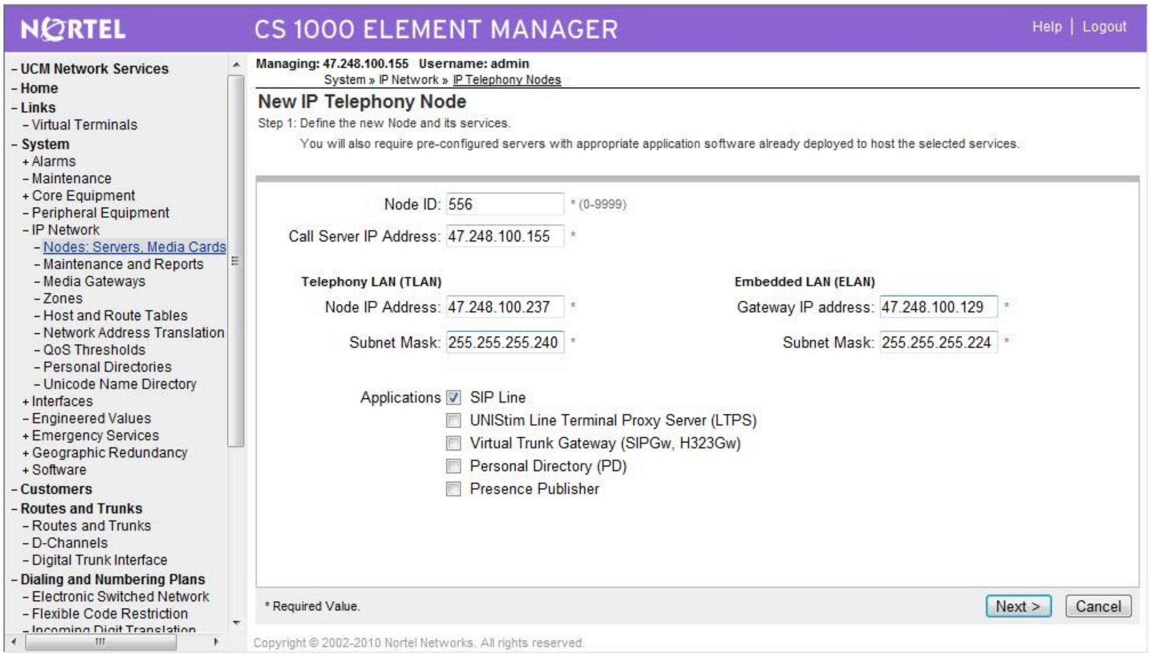

Figure 5 – IP Telephony Node

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

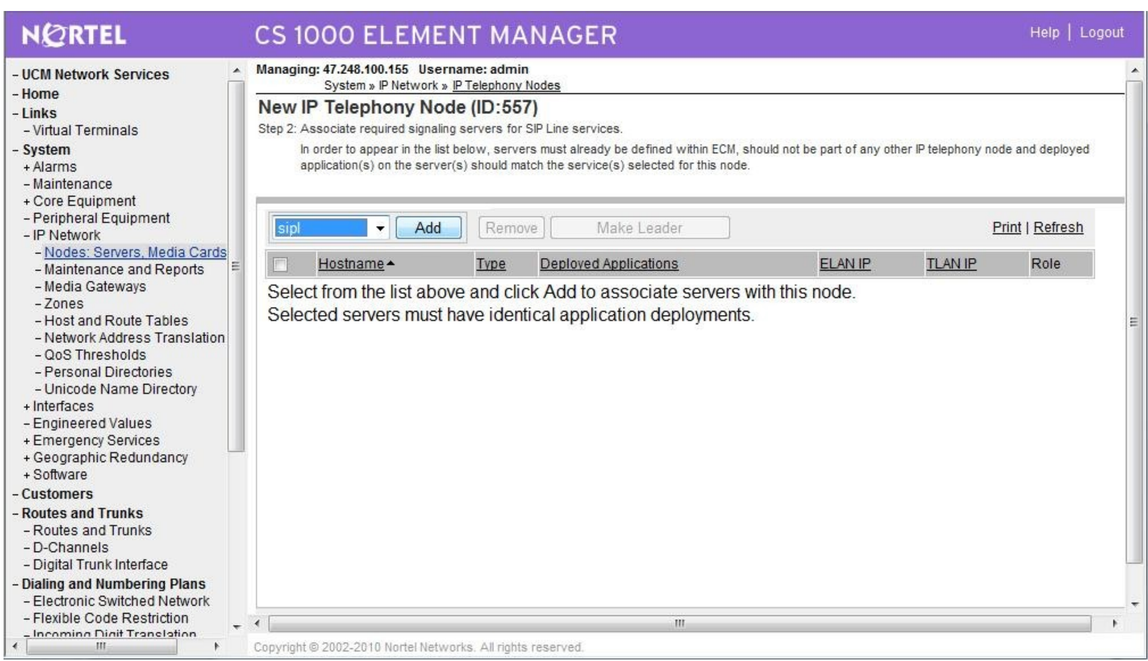

Figure 6 – IP Telephony Node – Add Server

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- Click ***Next.*** The SIP Line Configuration Detail page appears as shown in figure 7.
- Enter SIP Line domain name in ***SIP Domain name*** text box. This must be the same as the domain name configured in ***Customers***.

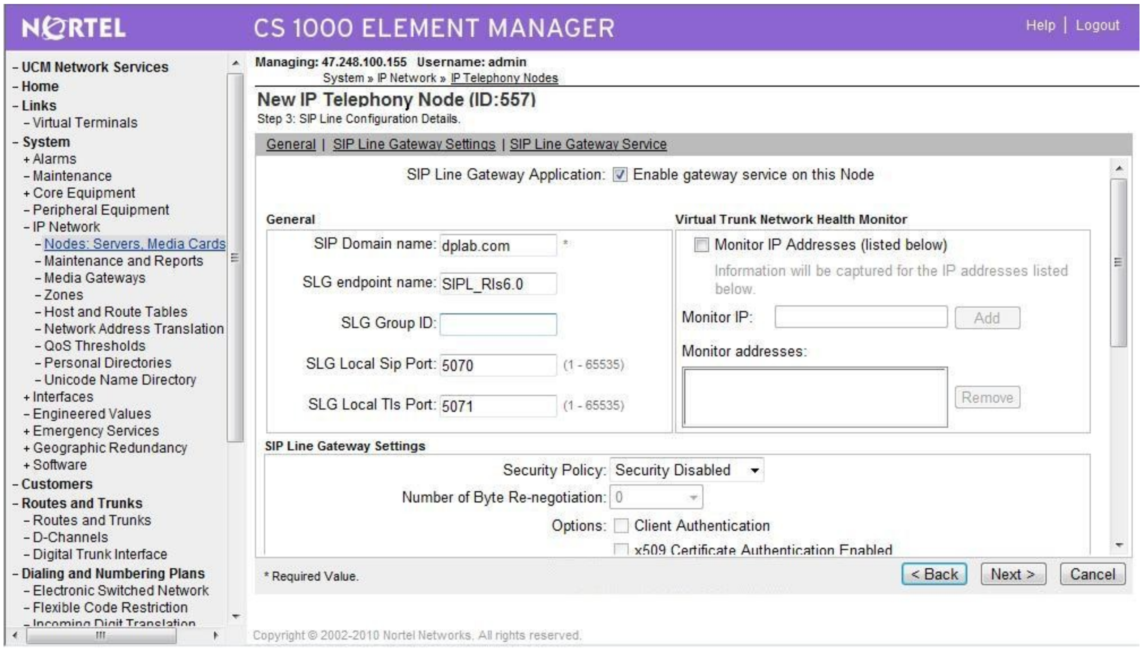

Figure 7 – SIP Line Node Details

- Under the ***SIP Line Gateway Services*** section, select ***MO*** from the ***SLG Role*** list.
- From the ***SLG Mode*** list, select ***S1/S2*** (SIP Proxy Server 1 and Server 2), figure 8.

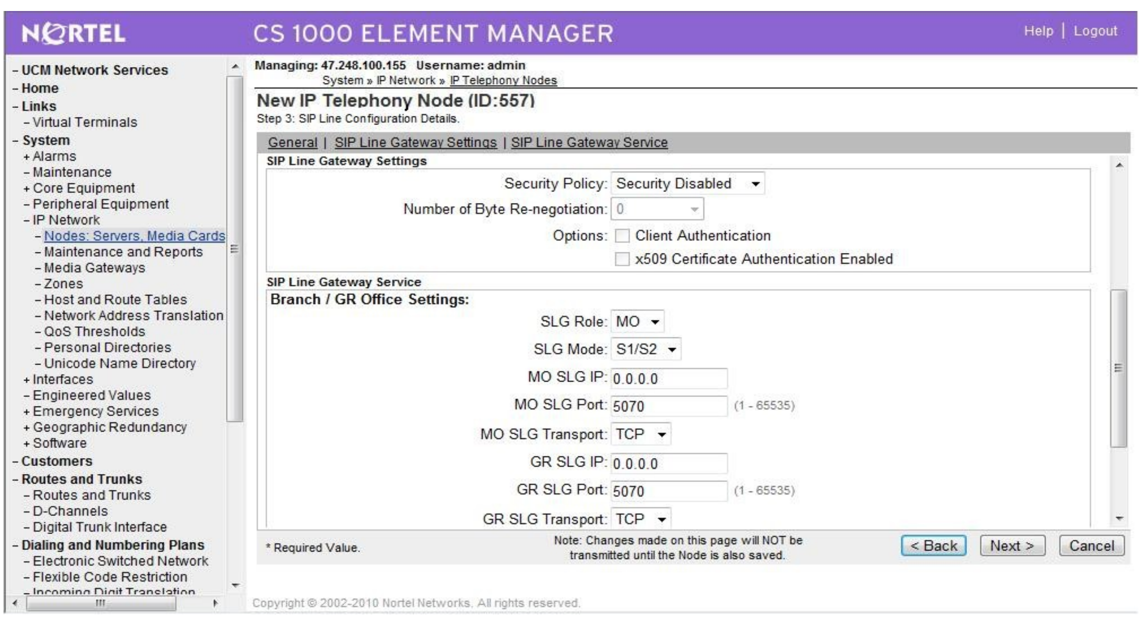

Figure 8 – SIP Line Node Details (cont.)

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

- Click ***Finish*** and wait for the configuration being saved. The ***Node Saved*** page appears, figure 9.

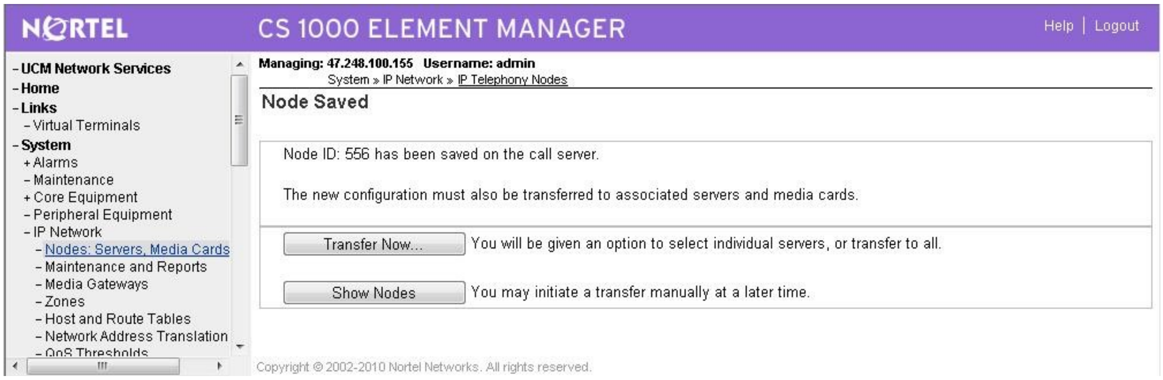

Figure 9 – Transfer Configuration

- Click ***Transfer Now.*** The ***Synchronize Configuration Files (Node ID 556) page*** appears.
- Select some or all of the node elements and then click ***Start Sync*** to transfer the configuration files to the selected servers, figure 10.

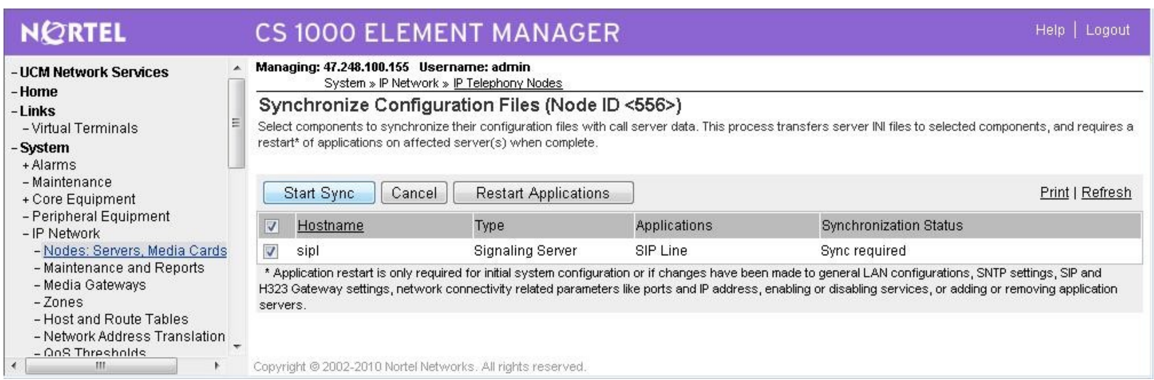

Figure 10 – Synchronize Configuration Files

4.5. D-Channel over IP Configuration

- On the EM page, on the left column menu navigate to ***Routes and Trunks → D-Channels.***
- Under the ***Configuration*** section, from the ***Choose a D-Channel Number*** list, select a D-Channel number, channel 30 in this configuration.
- Under the **Configuration** section, from the ***Type*** list, select ***DCH.***
- Click to ***Add.***
- From the ***D channel Card Type (CTYP)*** list, select ***D-Channels is over IP (DCIP).***
- Click to ***Add***.
- The ***D-Channels xx Property Configuration*** page appears as shown in figure 11.
- From the ***Interface type for D-channel (IFC)*** list, select ***Meridian Meridian1 (SL1)***.

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- Others are at default values.
- Click the ***Basic options (BSCOPT)*** link. The ***Basic options (BSCOPT)*** list expands.

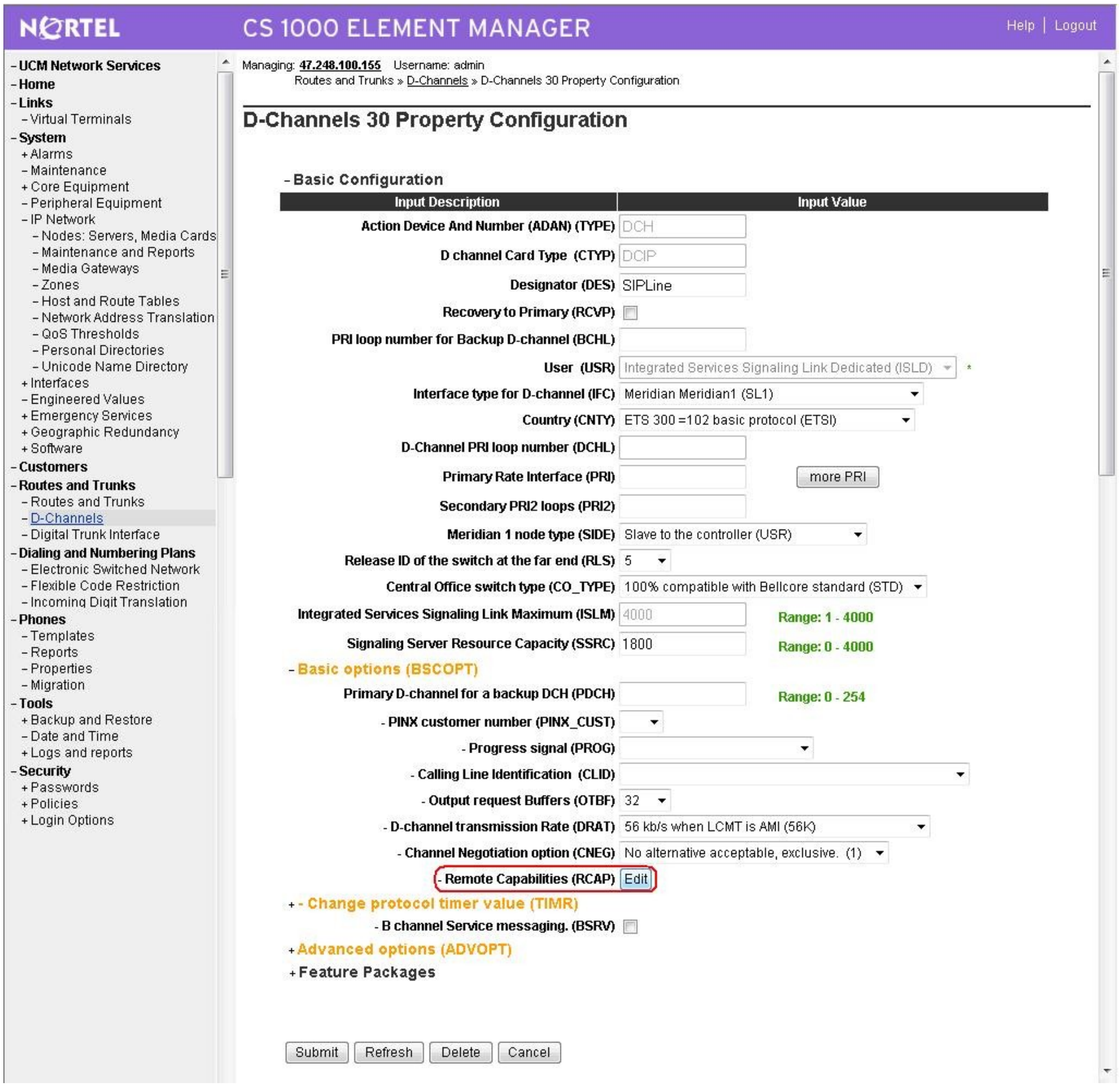

Figure 11 – SIP Line D-Channel Property Configuration

- Click ***Edit*** to configure ***Remote Capabilities (RCAP). The Remote Capabilities Configuration detail page*** will appear as shown in figure 12.
- 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***.
- The ***D-Channel xx Property Configuration*** page reappears.

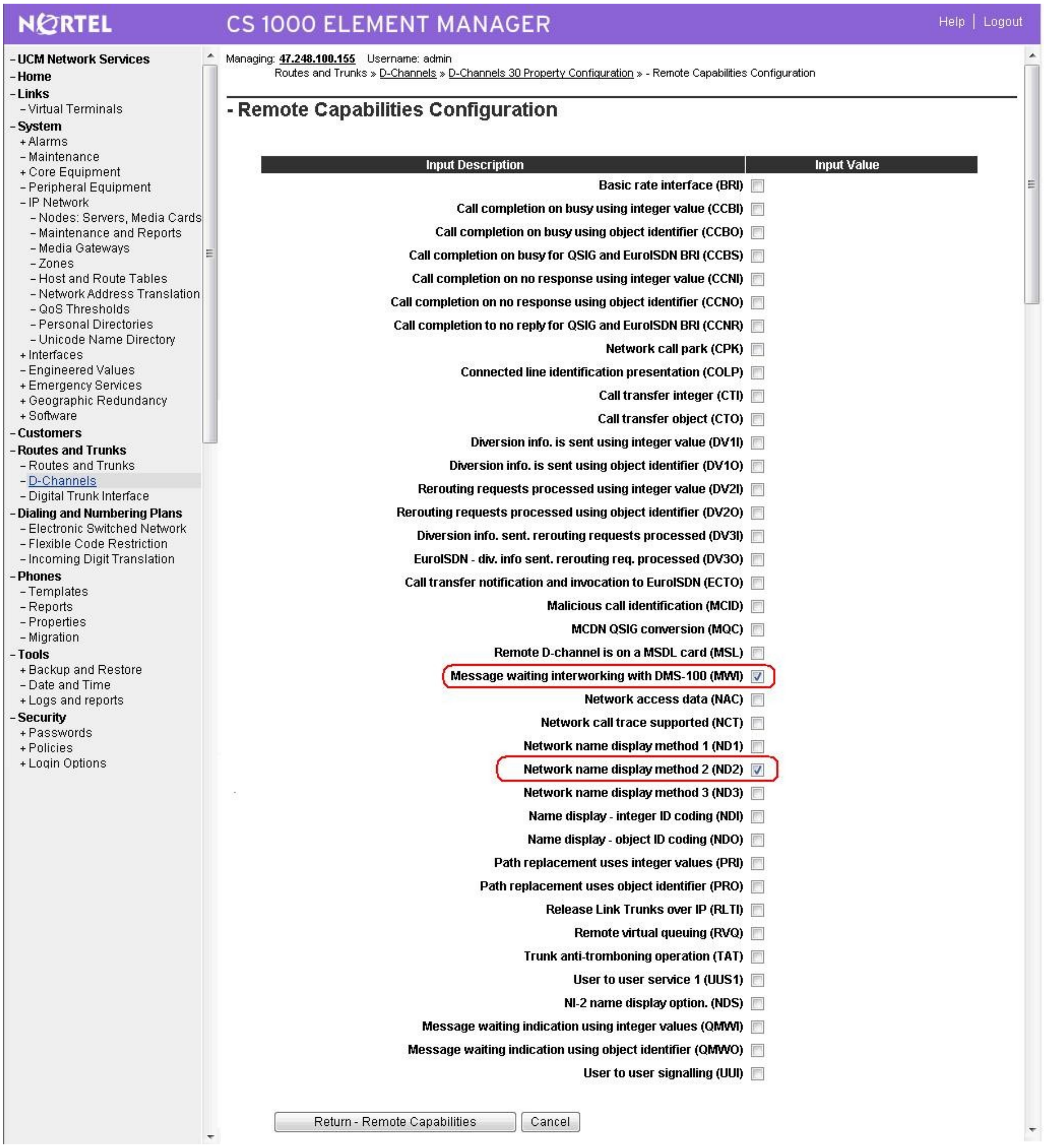

Figure 12 – SIP Line D-Channel RCAP Configuration Details

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.

Others check boxes are left unchecked.

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

- On the EM page, navigate to ***System → Interfaces → Application Module Link.***
- Click ***Add*** to add an Application Module Link. ***New Application Module Link*** page appears as shown in figure 13.
- Enter AML port in the ***Port number*** text box. The SIP Line Service can use port 32 to port 127. In this case, SIP Line Service is configured to use port 32.
- Click ***Save*** to save the configuration.

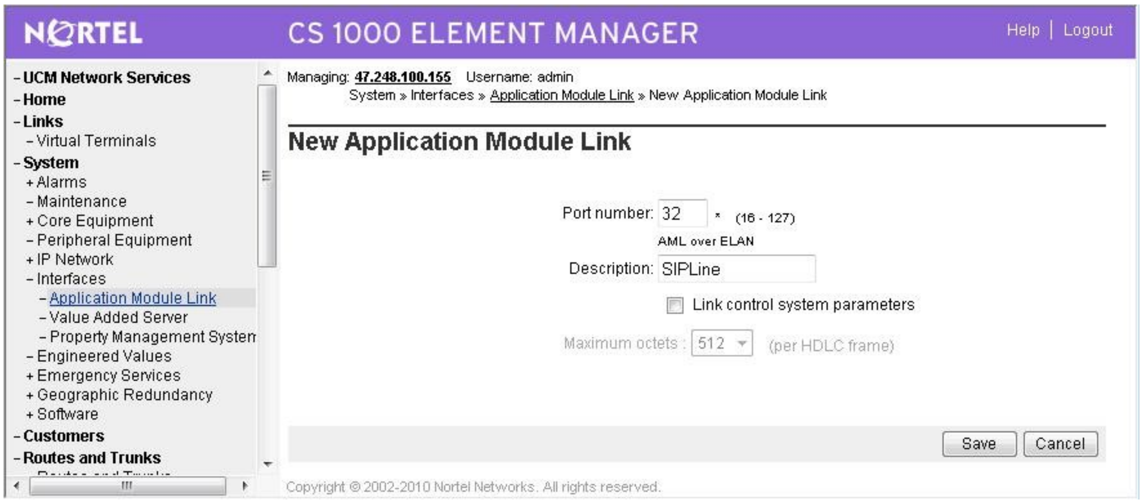

Figure 13 – Application Module Link Configuration

4.7. Value Added Server (VAS) Configuration

- On the EM page, navigate to ***System → Interfaces → Value Added Server.***
- Click ***Add*** to add new Value Added Server. The ***Add Value Added Server*** page appears.
- Click on the ***Ethernet LAN Link*** as shown in figure 14.
- Enter the Ethernet LAN Link number in the ***Ethernet LAN Link*** text box.
- Ensure that the ***Application Security*** check box is unchecked.

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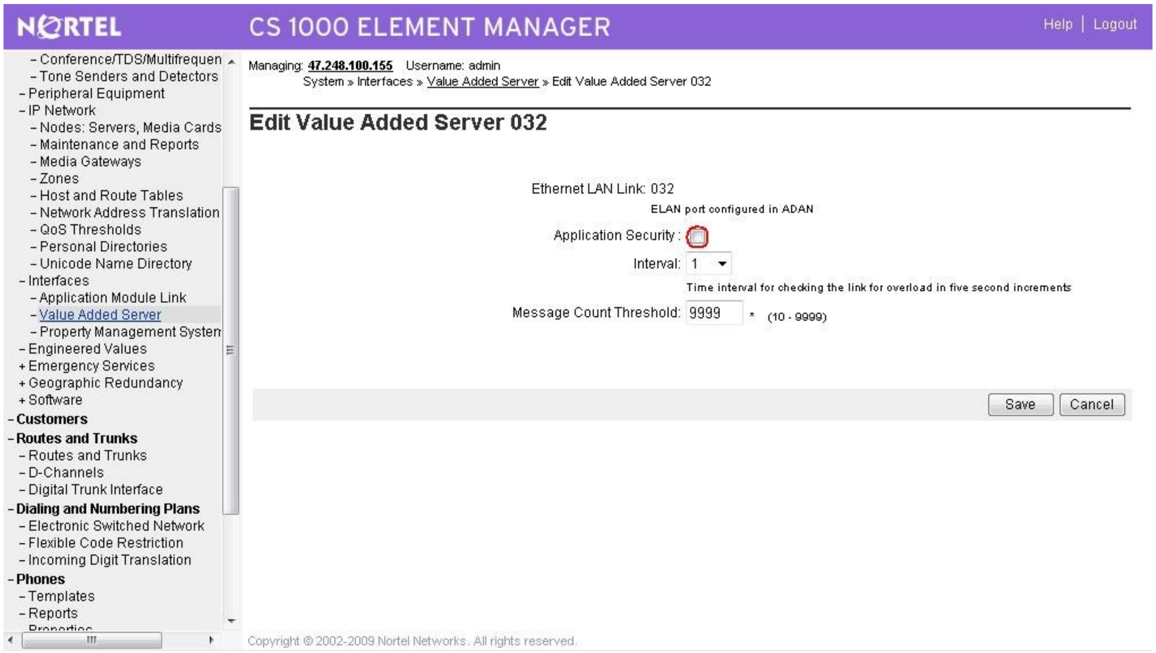

Figure 14 – Value Added Service for Application Module Link

4.8. Virtual Trunk Zone Configuration

- On the EM page, navigate to ***System → IP Network → Zones.***
- On the ***Zones*** page, select ***Bandwidth Zones***.
- On the ***Bandwidth Zones*** page, select a ***Bandwidth Zone number*** from the list, and click to ***Add*** (not shown).
- On the ***Zone Basic Property and Bandwidth Management*** page, set the zone properties based on bandwidth availability. It is recommended to set the ***Zone Strategy*** to ***BestQuality (BQ)*** as shown in figure 15.
- From the ***Zone Intent (ZBRN)*** list, select ***VTRK (VTRK)***.
- Click ***Submit*** button at the bottom of the page to save and commit the changes.

Figure 15 – Virtual Trunk Zone Configuration

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4.9. SIP Line Route Data Block (RDB) Configuration

- On the EM page, navigate to ***Routes and Trunks → Routes and Trunks.***
- Click ***Add*** for the customer number.
- On the ***Customer xx, New Route Configuration*** page as shown in figure 16.
- From the ***Route number (ROUT)*** list, select a route number.
- From the ***Trunk type (TKTP)*** list, select ***TIE trunk data block (TIE).***
- When Trunk Type (TKTP) is selected, the following options appear:
    - Trunk type M911P (M911P)
    - The route is for a virtual trunk route (VTRK)
    - Digital trunk route (DTRK)
    - Integrated services digital network option (ISDN)
- From the ***Incoming and outgoing trunk (ICOG)*** field, select ***Incoming and Outgoing (IAO)***.
- In the ***Access code for the trunk route (ACOD)*** field, enter the access code.
- Select ***The route is for virtual trunk route (VTRK)*** check box.
- In the ***Zone for codec selection and bandwidth management (ZONE)*** field, enter the zone number. (Use the same zone as configured in ***5.7 "Virtual Trunk Zone Configuration"***)
- In the ***Node ID of signaling server of this route (NODE)*** field, enter the node ID of the SIP Line Gateway.
- From the ***Protocol ID for the route (PCID)*** list, select ***SIP Line (SIPL).***
- Select the ***Integrated services digital network option (ISDN)*** check box.
- From the ***Mode of operation (MODE)*** list, select ***Route uses ISDN Signaling Link (ISLD)***.
- In the ***D channel number (DCH)*** field, enter the D-channel number.
- From the ***Interface type for route (IFC)*** list, select ***Meridian M1 (SL1)***.
- Ensure the ***Network calling name allowed (NCNA)*** check box is selected.
- Select the ***Trunk route optimization (TRO)*** check box. (Optional)
- ***Basic Route Options, Network Options, General Options, and Advanced Configurations*** sections set as default
- Click ***Submit button to save the configuration changes***.

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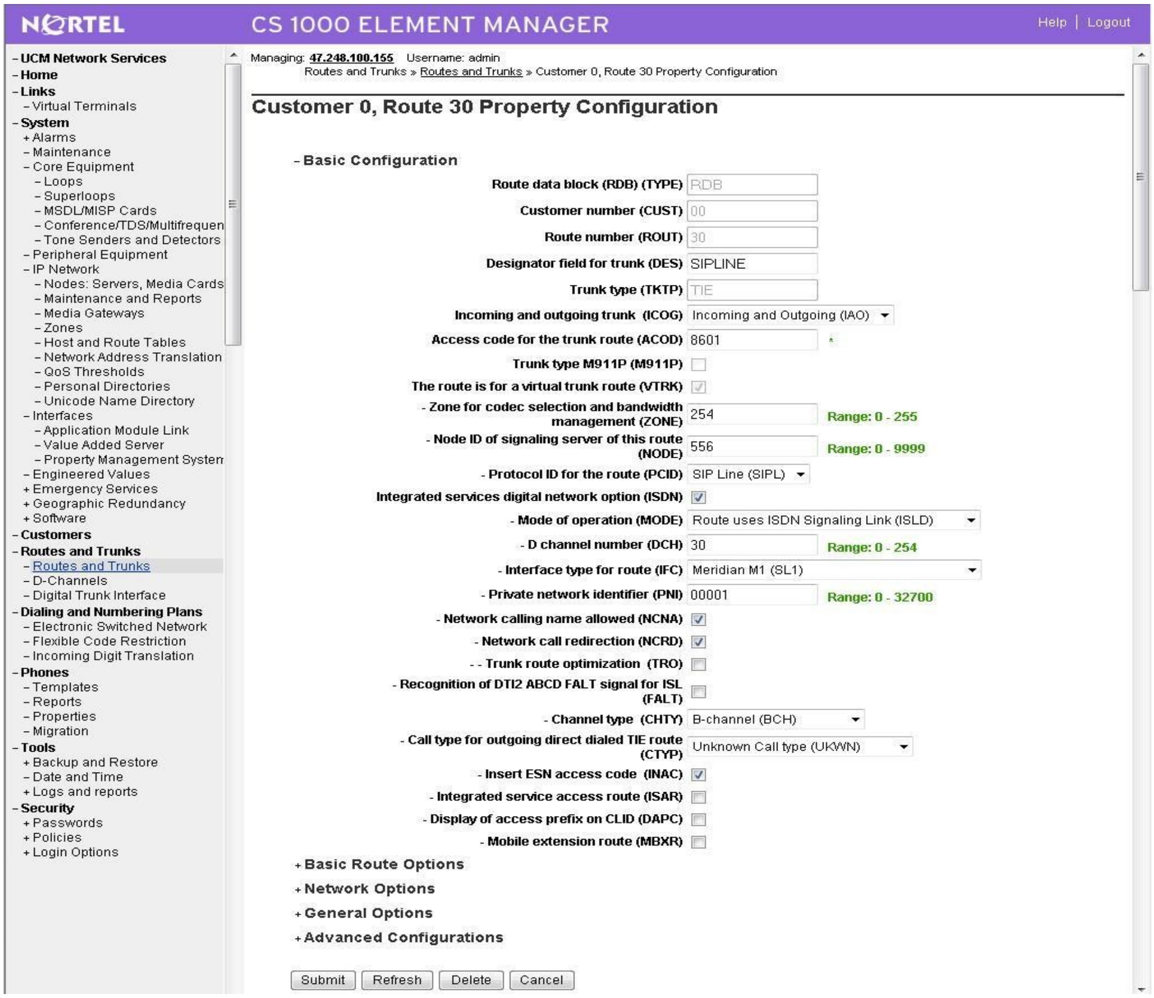

Figure 16 –SIP Line Route Configuration

Note: There is an outstanding issue (Q02073088) with the CS 1000 Call Waiting feature which occurs when ***Network Call Redirection*** is enabled. If the Network Call Redirection feature is not required, uncheck the feature to make the Call Waiting work.

4.10. SIP Line Virtual Trunk Configuration

- On the EM page, navigate to ***Routes and Trunks → Routes and Trunks.***
- Select the customer for which you are configuring Virtual Trunks.
- Click ***Add trunk associated with the route listing*** to add new trunk members.
- The ***Customer xx, Route yy, New Trunk Configuration*** Web page appears as show in figure 17.
- Choose ***Multiple trunk input number (MTINPUT)*** if you are using more than one trunk.
- From the ***Trunk data block (TYPE)*** list, select ***IP Trunk (IPTI)***.
- In the ***Terminal Number (TN)*** field, enter a TN.
- Enter a ***Route number***, ***Member number (RTMB)***.
- Enter a ***Trunk Group Access Restriction (TGAR)*** value.

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- In the ***Channel ID for this trunk (CHID)*** field, enter a ***channel ID*** (where the range is 1 to 382).
- To specify a ***Class of Service (CLS)*** for the trunk, click ***Edit***. The ***Class of Service Configuration*** Web page appears.
- Select a ***Class of Service***.
- Click ***Return Class of Service*** to return to the ***New Trunk Configuration*** Web page.
- Select ***Basic Configuration***. The ***Basic Configuration*** list expands.
- From the ***Start arrangement Incoming (STRI)*** list, select a value for the start arrangement for incoming calls.
- From the ***Start arrangement Outgoing (STRO)*** list, select a value for the start arrangement for outgoing calls.
- Select ***Advanced Trunk Configurations***. The ***Advanced Trunk Configurations*** list expands.
- Configure ***Network Class of Service group (NCOS).***
- Click ***Save***.

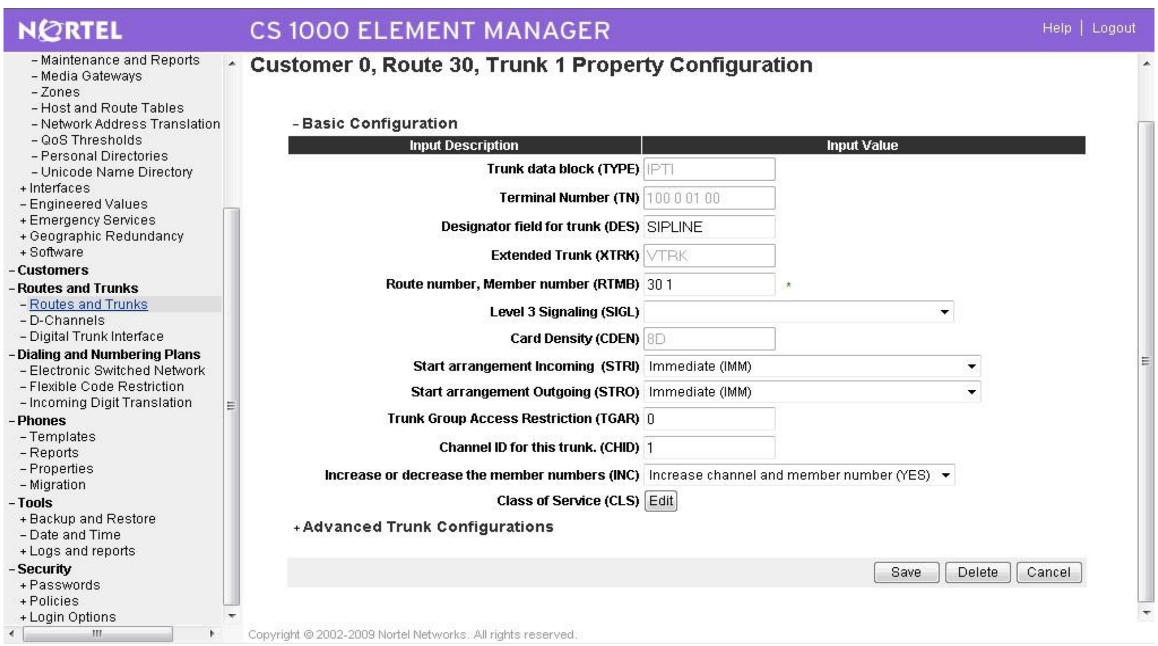

Figure 17 –SIP Line Trunk Configuration

4.11. SIP Line Phones Configuration

Following is a sample configuration for a Third Party SIP Line endpoint. Depending on supported features and service access level of the user, this configuration can be adjusted accordingly. The sample is using Command Line Interface of CS1000. This can be done by login to Call Server of CS1000 and using overlay 11 as shown below. The red text values are the changes required where others are at default values.

```
>LD 11

REQ: prt
TYPE: tnb
TN   96 0 1 27
DATE
PAGE
DES

DES MEDIA5
TN   096 0 01 27 VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 0← Set this to 1 and set SIP3 to 0 if this TN is reserved for Avaya SIP Phones
SIP3 1 ← Set this to 1 and set SIPN to 0 if this TN is reserved for third party SIP Phones
FMCL 0
TLSV 0
SIPU 55573
NDID 556
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 001
CUR_ZONE 001
ERL
ECL  0
FDN  55576    ← If CLS FNA is equipped, call will be forwarded no answer to this number
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 2        ← This field must be set first if call pickup is equipped (CLS PUA)
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
```


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MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD **SWD** LND **CNDA**
CFTD SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUA DPUA **DNDA** CFXA ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHA IPND **DDGA NAMA** MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3
MCBN
FDSD NOVD VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH
CPND_LANG ENG
RCO 0
HUNT **55576** ← If CLS HTA/FBA is equipped, call will be forwarded busy to this number
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 **SCR 55573** 0 MARP
CPND
CPND_LANG ROMAN
NAME **Media5 55573**
XPLN 13
DISPLAY_FMT FIRST, LAST
01 HOT U **2655573** MARP 0
02 SCU **0004** ← Speed Call User
03
04 **MSB** ← This key can be different than key 04 to enable Make Set Busy (MBS) feature
05
.
.
16
17 TRN
18 AO6
19 CFW 16 55574
20 RGA
21 PRK
22 RNP
23

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24 PRS
25 CHG
26 CPN

4.12. PSTN Outside Trunk Configuration

Following is a sample configuration which was used during compliance testing. For more information about PRI Trunk Configuration, see **NN40031-301 Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning.**

4.12.1. Procedure summary

This procedure is applied for CS 1000 systems under test. Please refer to Figure 1. These provisioning are using Command Line Interface by login to Call Server of CS1000.

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

4.12.2. Adding a PRI card

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

Prompt	Response	Description

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

4.12.3. Adding a PRI D-channel

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

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	CFN	Configuration data block.
ADAN	NEW DCH 10	Add a primary D-channel (any unused SDI port.) xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4. Xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
CTYP	TMDI	Card type where: MSDL = The NTBK51BA Downloadable D-Channel Daughterboard. TMDI = TMDI (NTRB21) card.
DES	T1_QSIG	Designator field.
USR	PRI	D-channel is for ISDN PRI only.

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		Note: 2.0 Mb only supports PRI or SHA user
IFC	ISGF	Interface type.
DCHL	10	PRI card number carries the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above. Where: xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4. xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
SIDE	NET	NET = network, the controlling switch (applied for CS 1000 PSTN simulator USR = slave to the controller (applied for CS 1000 system under test)
RLS	6	Software release of far-end. This is the current software release of the far-end. If the far-end has an incompatible release of software, it prevents the sending of application messages, for example, 'Network Ring Again.
RCAP	CCBI CCNI PRI DV3I CTI QMWI	Remote Capabilities.
PR_TRIGS PR_TRIGS PR_TRIGS PR_TRIGS	DIV 2 3 CNG 2 3 CON 2 3 CTR2 2 3	Path Replacement Triggers

4.12.4. Defining a PRI customer

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

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer data block.

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CUST	0	Customer number.
ISDN	YES	Customer is equipped with ISDN.

4.12.5. Defining a PRI service route

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

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

4.12.6. Defining service channels and PRI trunks

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

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

5. Configure Mediatrix 4104

This section describes how to access the M4104 gateway web interface and configure the M4104 VOIP gateway for testing.

5.1. SIP Registration

In the web browser address field, enter the M4104 gateway IP address. The M4104 login page will appear as shown in figure 18. Enter the user name and password.

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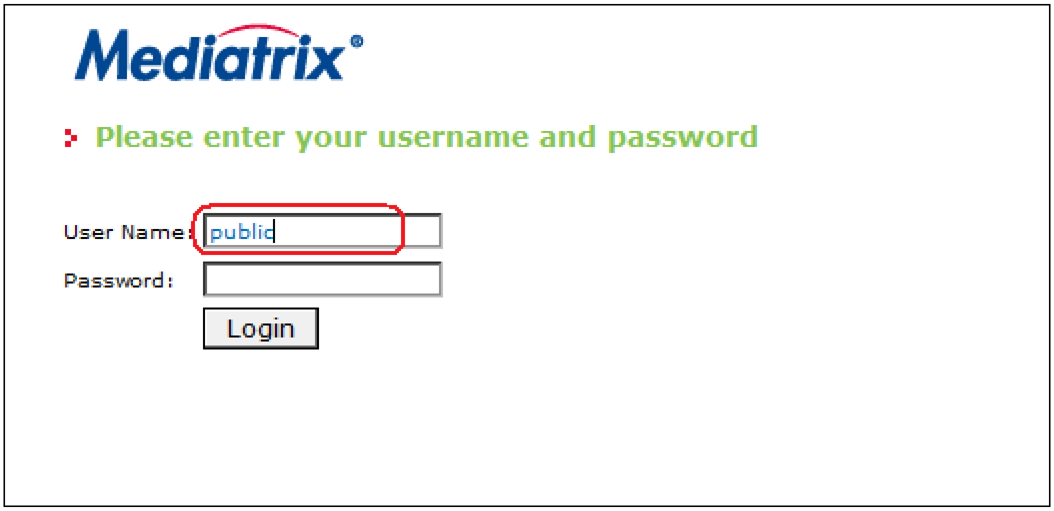

Figure 18 – Login Screen

Click the Login button, the main configuration screen appears as in figure 19 below.

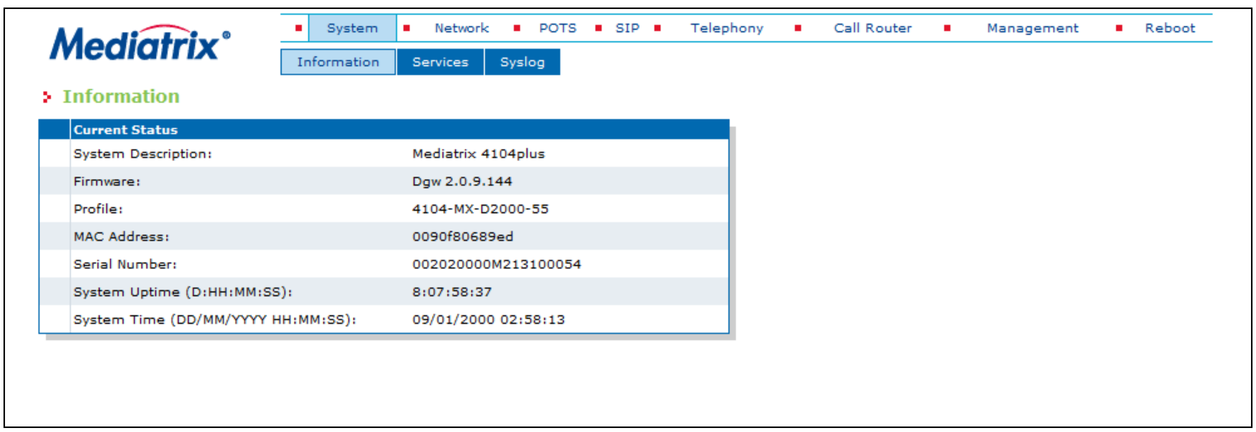

Figure 19: Main Configuration Screen

5.2. Configure the SIP Port

To configure the SIP Port, in the main configuration screen (see figure 19), click SIP menu on the top menu bar, and then click on Gateways (see figure 20 below).
In the SIP Port field, type the SIP port number. This example uses 5070.
Click Submit.

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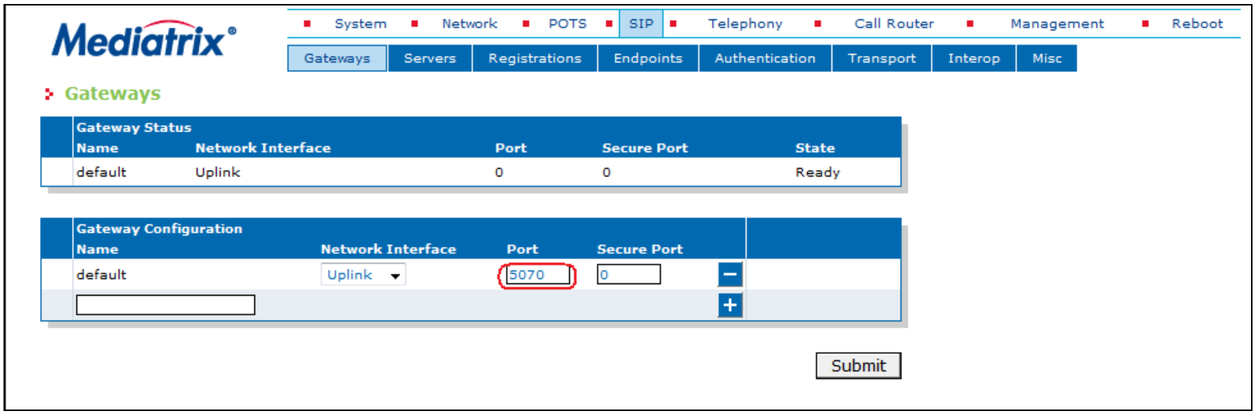

Figure 20: Gateway Screen

After click Submit button on figure 20 above, user needs to restart a service for the change to take affect on the M4104 gateway. The next step is to click on Service table as indicated in figure 21.

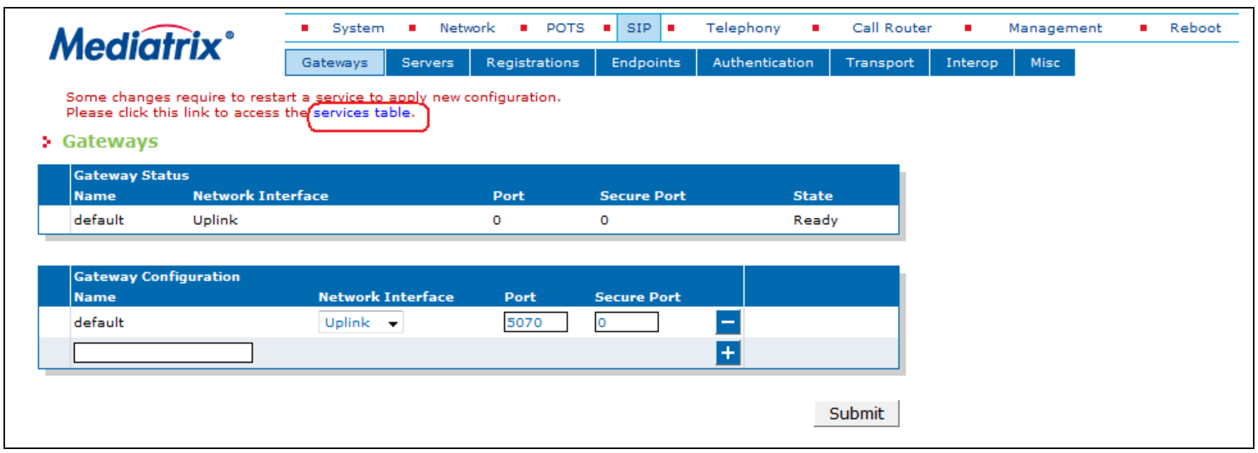

Figure 21: Restarting Services

The Services screen appears as shown on figure 22 below, click on Restart Required Services button to restart the SIP Endpoint service.

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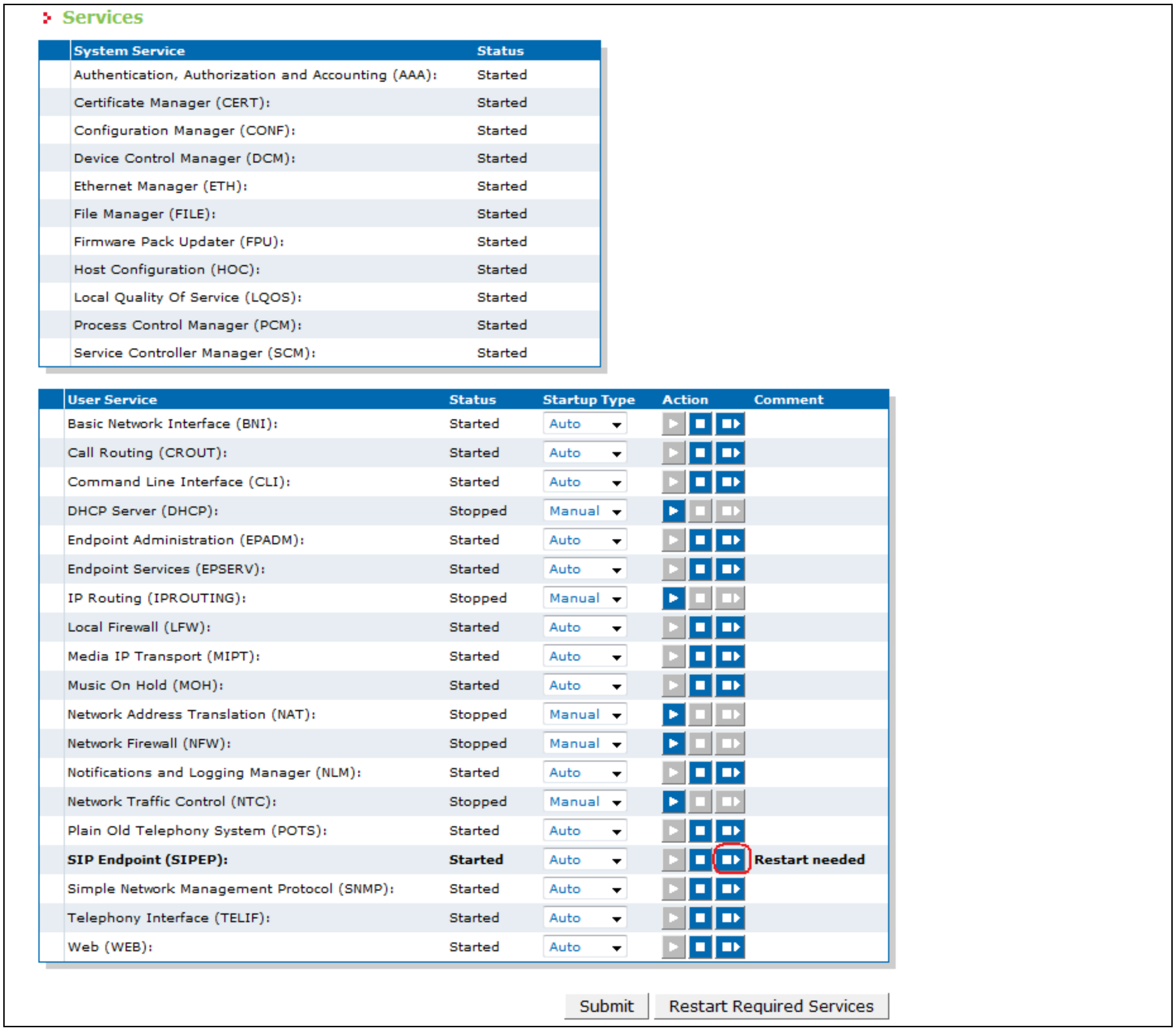

Services

System Service	Status
Authentication, Authorization and Accounting (AAA):	Started
Certificate Manager (CERT):	Started
Configuration Manager (CONF):	Started
Device Control Manager (DCM):	Started
Ethernet Manager (ETH):	Started
File Manager (FILE):	Started
Firmware Pack Updater (FPU):	Started
Host Configuration (HOC):	Started
Local Quality Of Service (LQOS):	Started
Process Control Manager (PCM):	Started
Service Controller Manager (SCM):	Started

User Service	Status	Startup Type	Action	Comment
Basic Network Interface (BNI):	Started	Auto		
Call Routing (CROUT):	Started	Auto		
Command Line Interface (CLI):	Started	Auto		
DHCP Server (DHCP):	Stopped	Manual		
Endpoint Administration (EPADM):	Started	Auto		
Endpoint Services (EPSERV):	Started	Auto		
IP Routing (IPROUTING):	Stopped	Manual		
Local Firewall (LFW):	Started	Auto		
Media IP Transport (MIPT):	Started	Auto		
Music On Hold (MOH):	Started	Auto		
Network Address Translation (NAT):	Stopped	Manual		
Network Firewall (NFW):	Stopped	Manual		
Notifications and Logging Manager (NLM):	Started	Auto		
Network Traffic Control (NTC):	Stopped	Manual		
Plain Old Telephony System (POTS):	Started	Auto		
SIP Endpoint (SIPEP):	**Started**	Auto		**Restart needed**
Simple Network Management Protocol (SNMP):	Started	Auto		
Telephony Interface (TELIF):	Started	Auto		
Web (WEB):	Started	Auto		

Submit | Restart Required Services

Figure 22: Services Screen

The confirmation screen will appear to let the user know that the service was restarted successfully as shown in figure 23 below.

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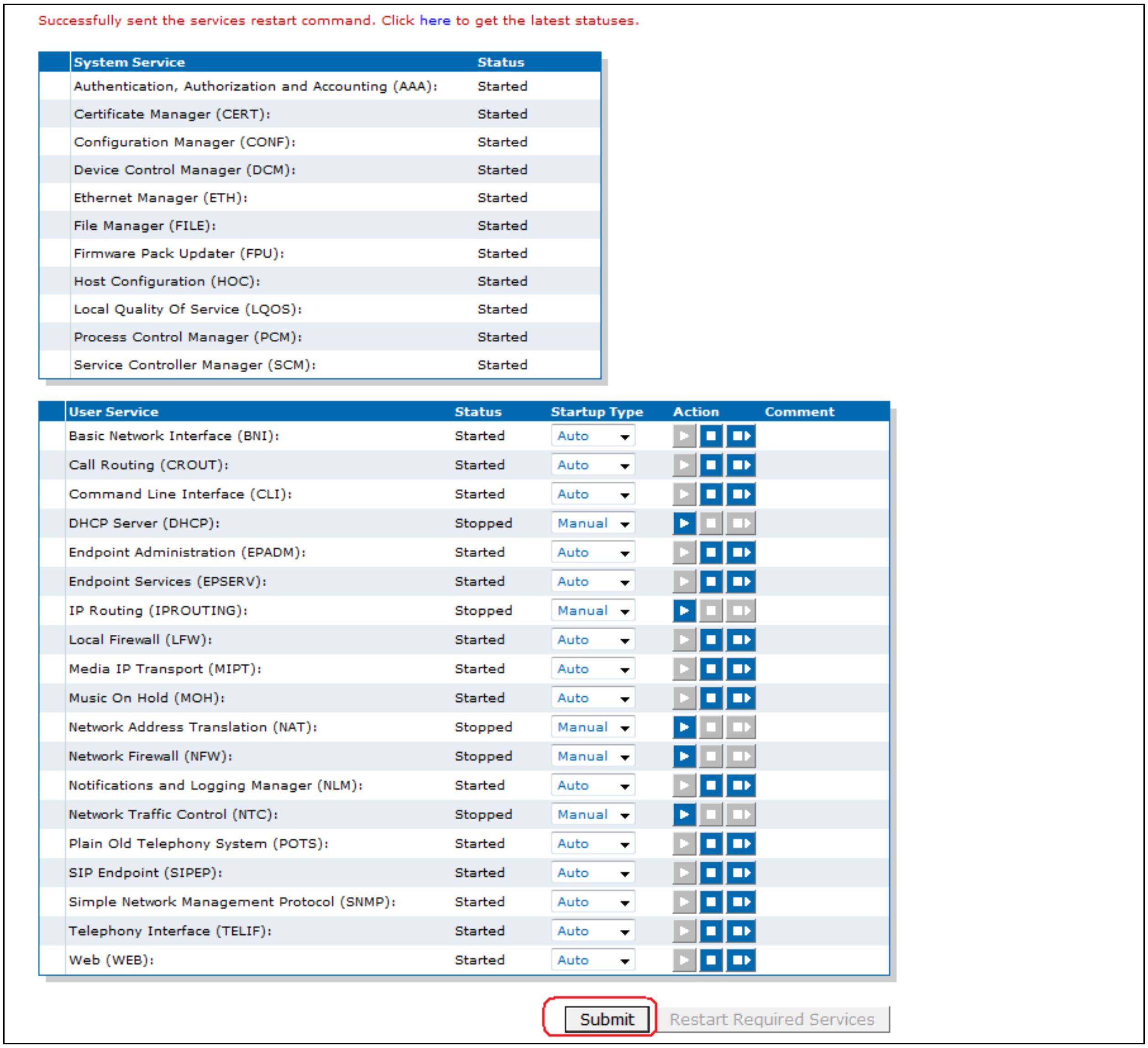

Figure 23

After click Submit button on figure 23, the correct SIP port appears in the Gateway option as shown in figure 24.

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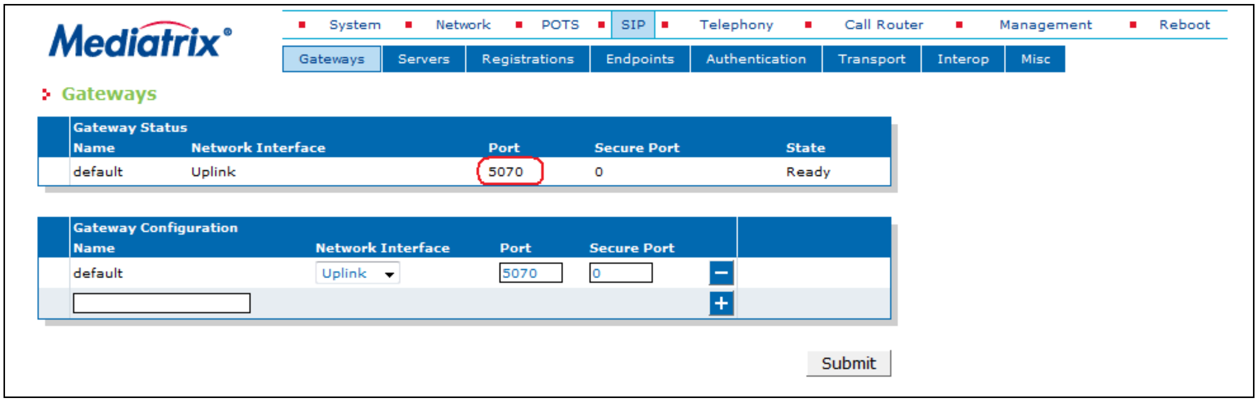

Figure 24: Confirming the SIP Port

5.3. Configure communication to the SIP Proxy Server

The next steps are to configure communication to the SIP proxy server that the Mediatrix4104 will use to route VoIP calls.

In the main configuration screen (see figure 19), click SIP, and then click Servers
The Servers screen appears as shown in figure 25.
In the Registrar Host and Proxy fields, enter the correct SIPLine Gateway IP address and SIP port (shown on page 8, figure 5 – Node IP address value, 47.248.100.237 and port 5070). Click Submit button to save the configured information.

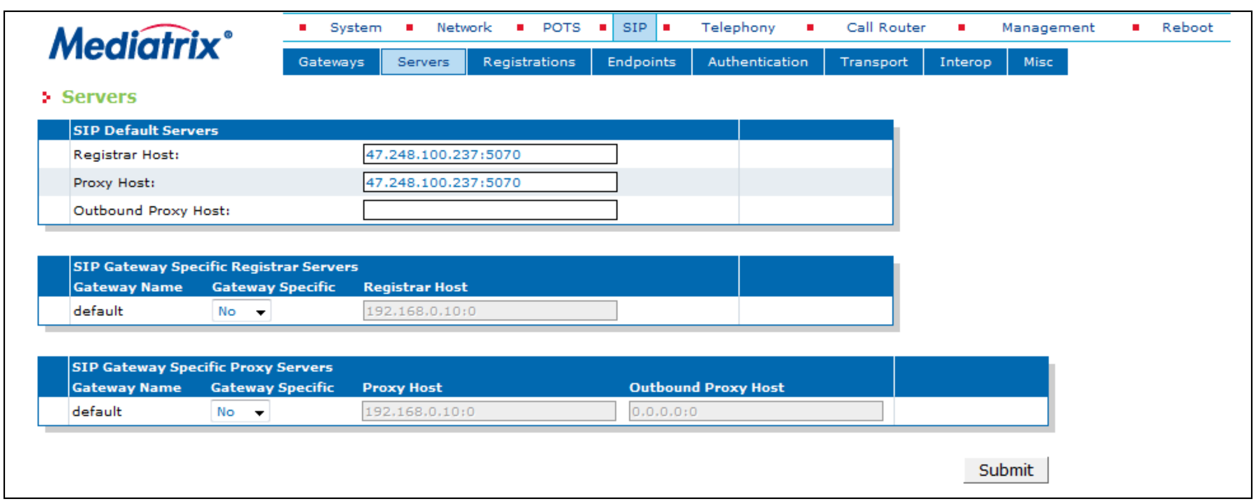

Figure 25: Servers Screen

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5.4. Configure Registrations and Authentications

In the main configuration screen (see figure 19), click SIP, and then click Registrations.
The Registrations screen appears as shown in figure 26.
In the Endpoints Registration, enter a User Name. The Friendly Name field is optional and can be left blank.
Enable the Endpoint by choosing Enable option under the register column from pull down menu.
Leave the Gateway Name field as default.
Click on Submit button to save the configuration changes.

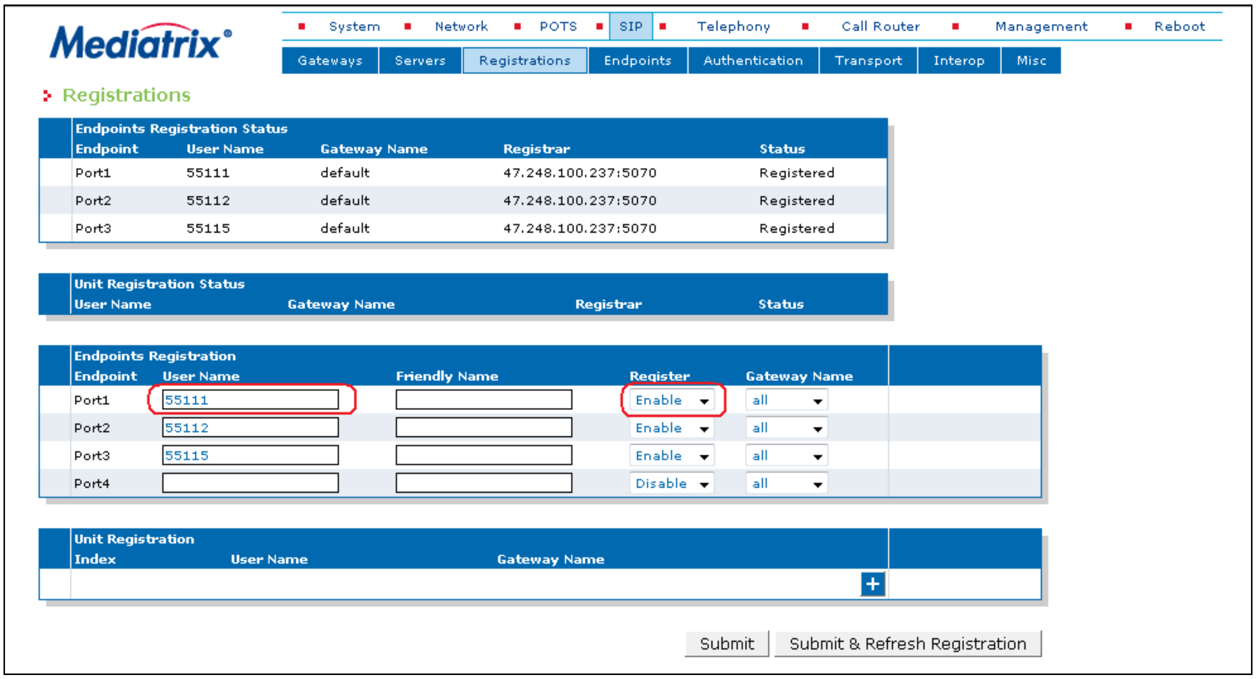

Figure 26: Registrations Screen

Continue to click on Authentications in the task bar, the Authentications screen appears shown as in figure 27.

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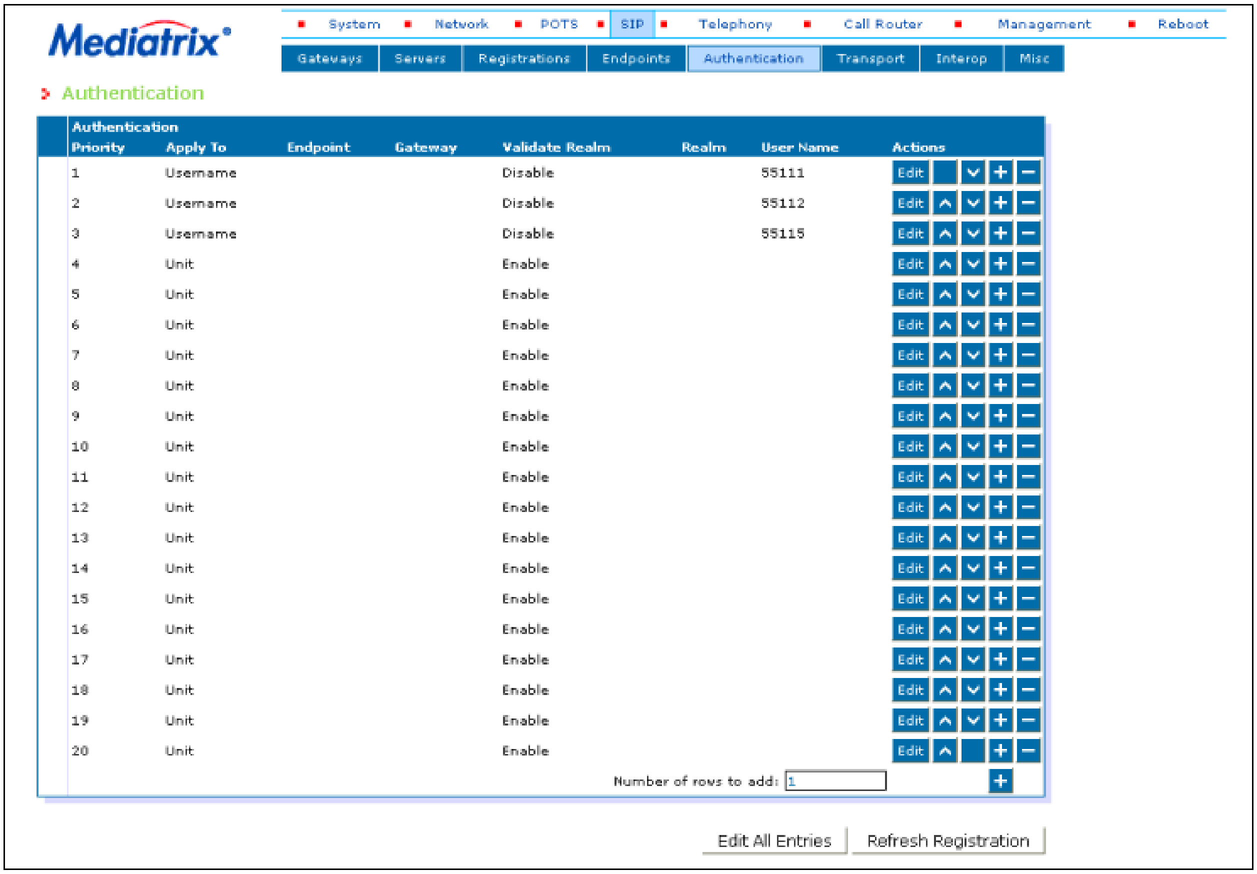

Figure 27: Authentications Screen

On the Authentication page, click on the Edit button to configure authentication method for each individual User as shown in figure 28.
Select Username from the "Apply To" drop down menu.
Select Disable from the Validate Realm list box.
Enter Username and Password.
Click Submit.

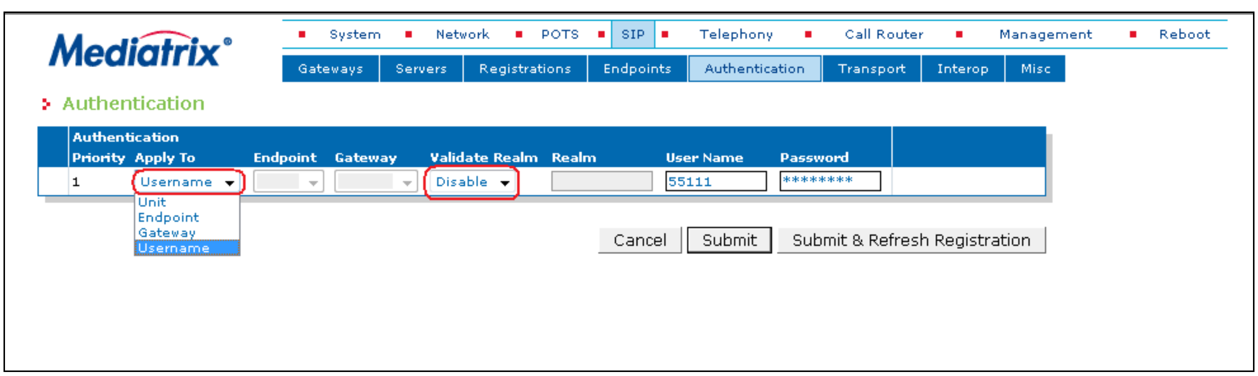

Figure 28

On the Misc page, configure the SIP Domain Override to the one required by the CS1000 (dplab.com) as shown in figure 29 below. This SIP line domain name is the one that has been created in section 4.4, referring to figure 7.

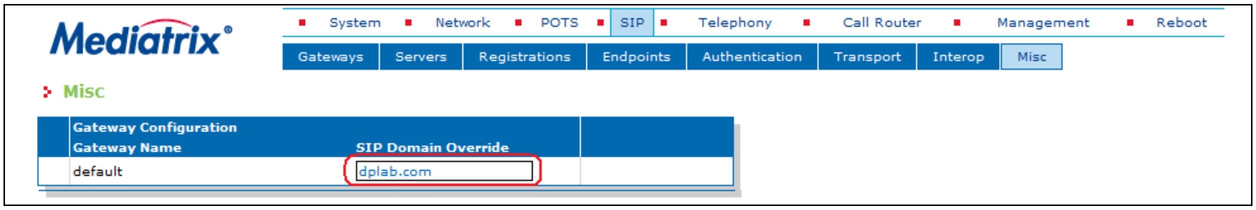

Figure 29

5.5. DTMF and Codec settings

This section describes how to configure DTMF and Codec settings in the M4104 gateway.

In the main configuration screen (see figure 30), click on the Telephony, and then click on CODECS menu from the second top menu bar
In the CODEC table list, ensure that G.711 a-Law; G.711 u-Law and G.729 are Enable in the Voice list.
In the Misc section, Level for Jitter Buffer is Normal; under DTMF Transport, select transport method from the Transport Method drop-down list as shown in figure 30 below.

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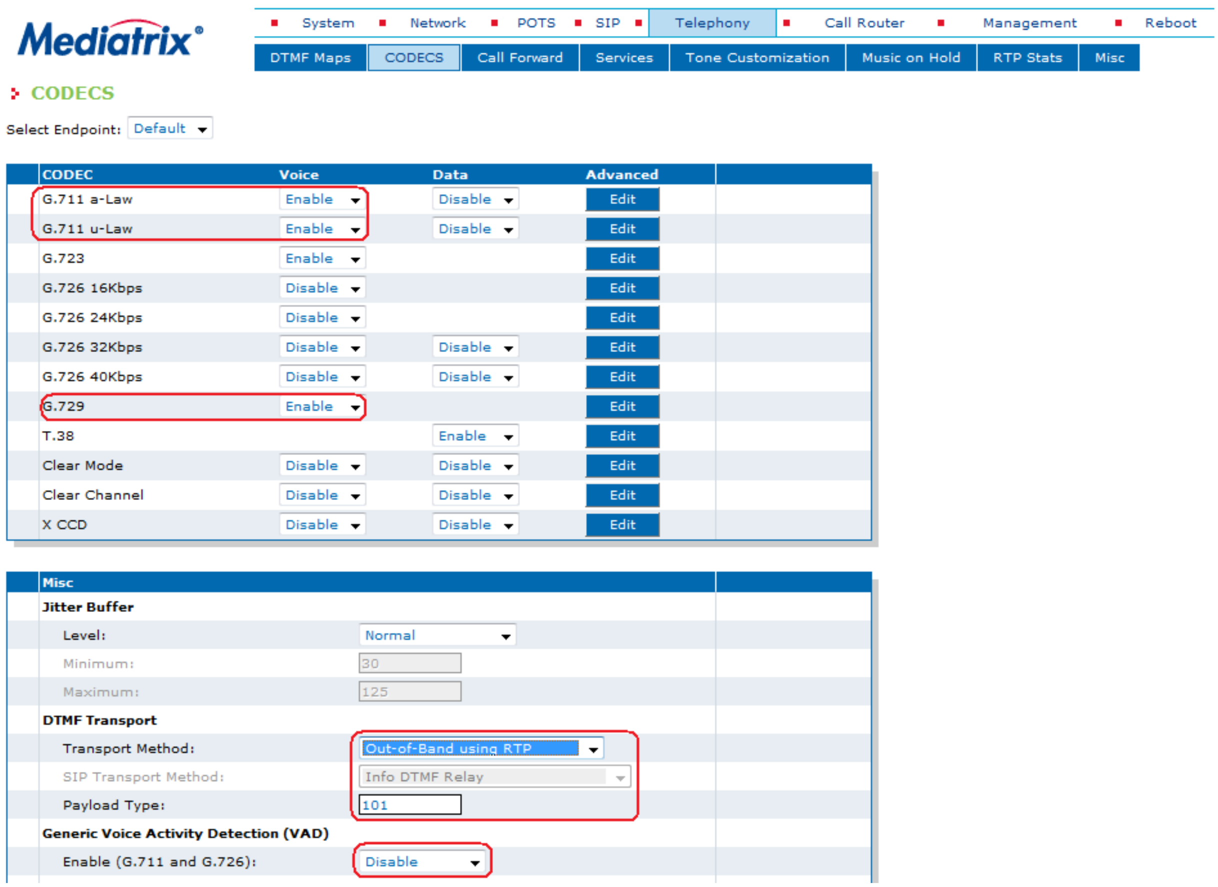

Figure 30 DTMF and Codecs setting

5.5.1. Disable G.729 Voice Activity Detection

Select the Telephony -> CODECS page.

Click on the Edit button in the Advanced column next to G.729 to modify the advanced settings of the CODEC as shown in figure 31.

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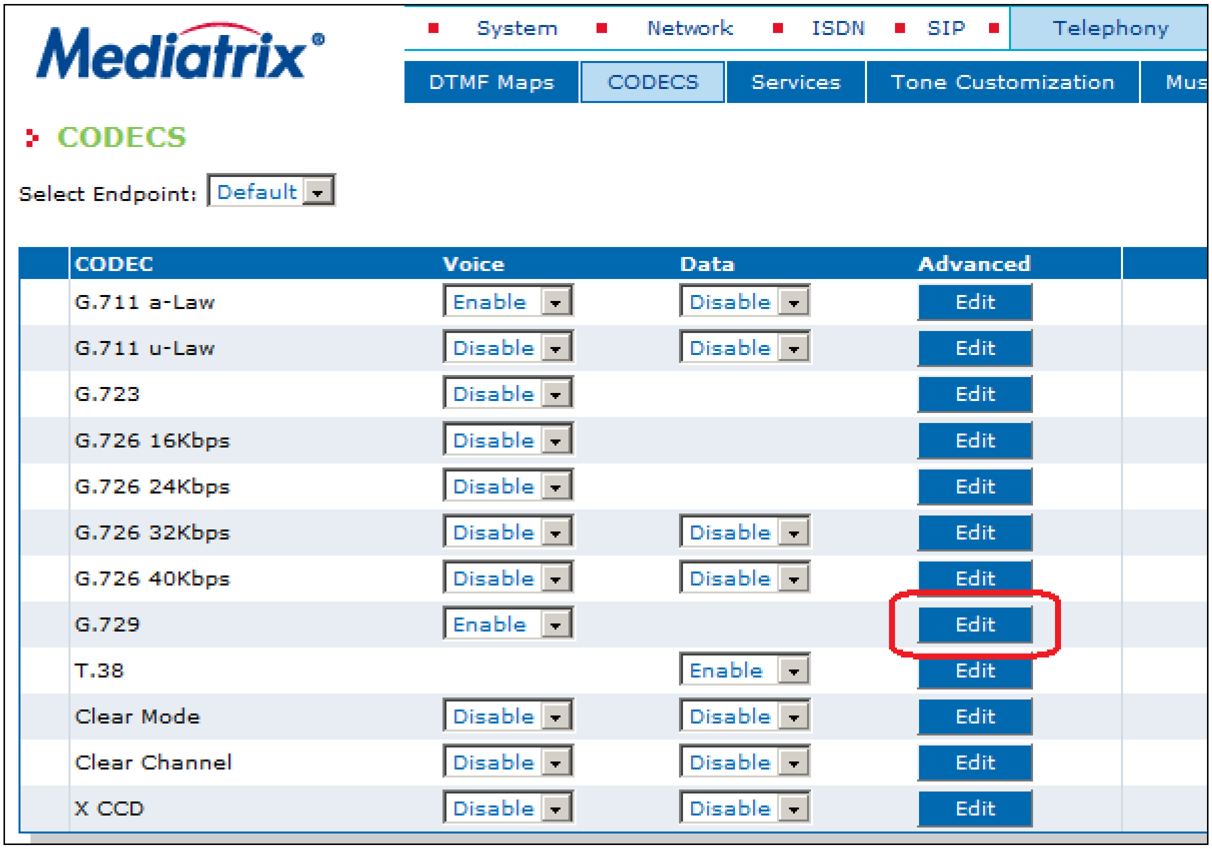

Figure 31

Set the Built-in Voice Activity Detection (VAD) to Disable and click on Submit button to accept the modifications as shown in figure 32 below.

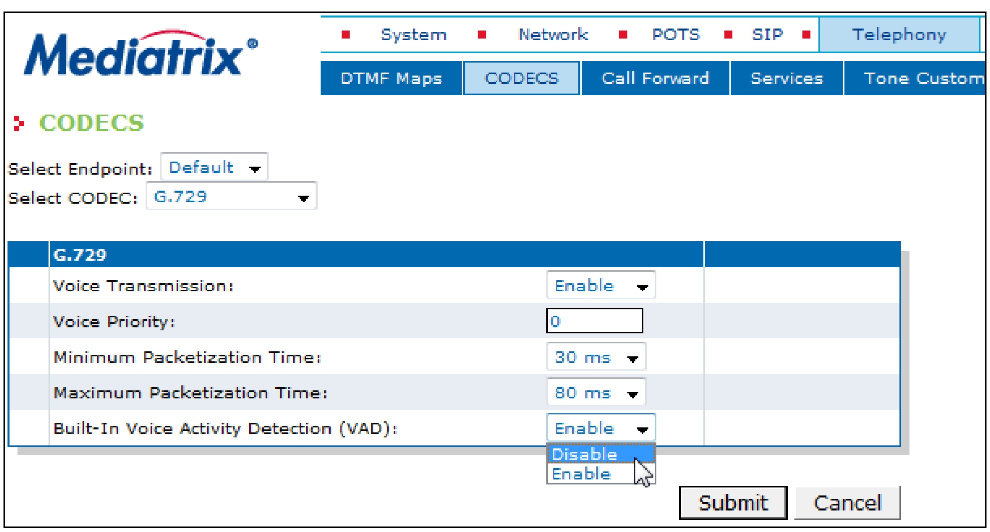

Figure 32

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5.5.2. Configuring DTMFs

Select the Telephony -> CODECS page, then scroll down to the section Misc.
Set the Transport Method to Out-of-Band using RTP.
Set the DTMF Transport Payload Type to 101.
Click the Submit button at the bottom right to accept the modifications, see figure 33.

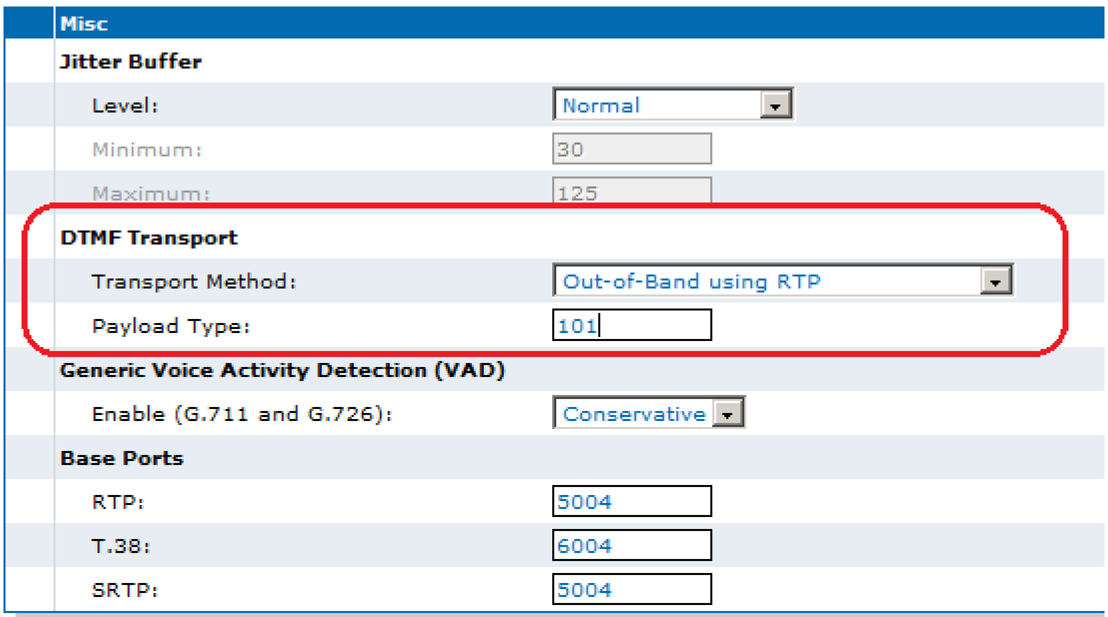

Figure 33

5.6. Advanced interoperability configuration

Unfortunately, not all configuration variables can be accessed via the web page. In order to access some less-used variables, the CLI (Command Line Interface) or a SNMP browsing tool, such as the Unit Manager Network from Media5 can be used. The steps below describe how to configure these variables using Unit Manager Network (called from now on UMN).

Go to the link below to download application
http://www.media5corp.com/en/support-a-training

On the support web page, click on the Go button in the Download & Documentation
In the Select Product Line drop down box, select the UNM v3.2. This is under the Software Application as shown in figure 34 below.

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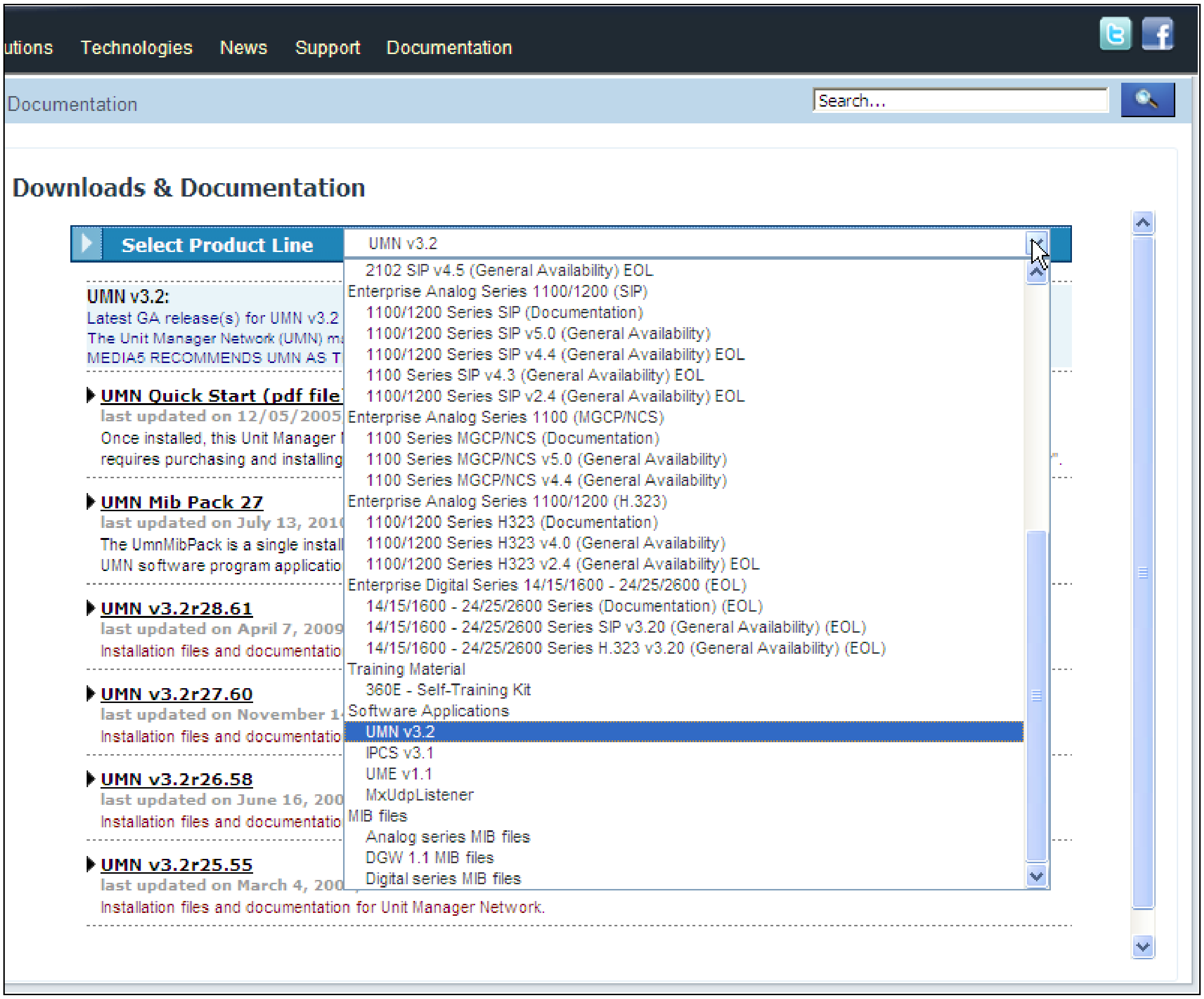

Figure 34

Download the UMN v3.2.28.61 as well as the UMN MIB Pack 27. Install UMN on the PC following the instruction and restart PC, once it asks for it. Once the PC restarted, install the UMN MIB Pack 27.

5.6.1. Adding a Unit in Unit Manager Network

The Unit Manager Network (UMN) software is a configuration and management tool for Mediatrix devices. The UMN is provided on the Media5 download portal. It has a default 3-units limit upon installation. This will suffice for most configuration deployment

Please refer to the UMN Quick Start guide for the installation of the software. Once the UMN software has been installed on your PC, proceed with the following steps.

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5.6.2. Start the UMN

Select from the Start Menu > Programs > Unit Manager Network 3.2 > Unit Manager Network.

5.6.3. Login to the UMN

In the *Administrator* login window (Connect to Unit Manager), a User Name and Password are not required. Click OK to proceed.

5.6.4. Perform an Autodetect

On the left pane, right-click the *Unit Manager* level, then select Autodetect. (see figure 35)

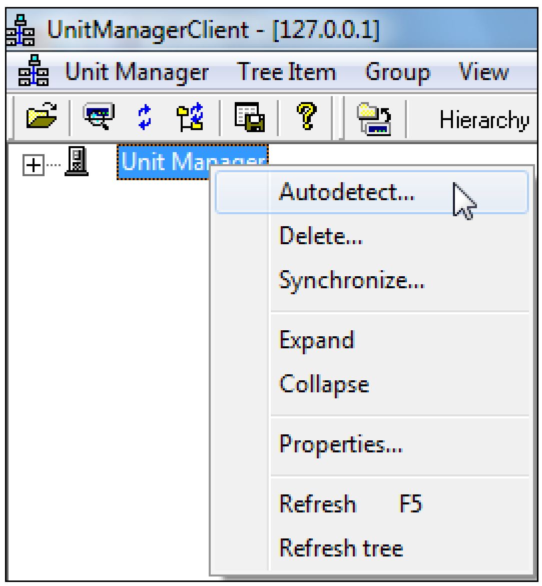

Figure 35

- Set the *IP Address Range* to minimize the time it takes to auto-detect the unit. Click **Start** to begin Mediatrix unit detection. When the unit is detected, the *Result* section lists the unit. (see figure 36)
- Select the unit and click OK. If no DHCP server is used in your subnet, you must connect one unit at a time since they will start by using the default IP address 192.168.0.1 after a recovery reset. You will have to set a different static IP address for every unit. Please see your unit's SIP Quick Start Guide for more details on initial setup.

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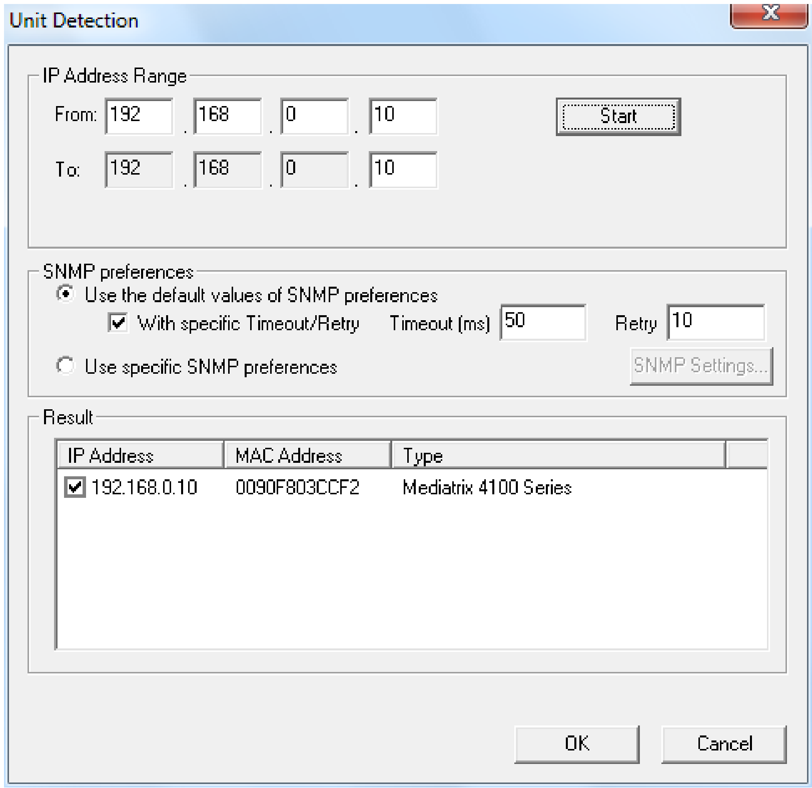

Figure 36

5.6.5. Advanced configuration

For every variable that are required to modify, follow this procedure:

- Select your Mediatrix gateway and right-click on the unit. Select Edit SNMP, as shown in figure 37.
- Make sure you click "Automatic Get" at the top, as shown in figure 38.

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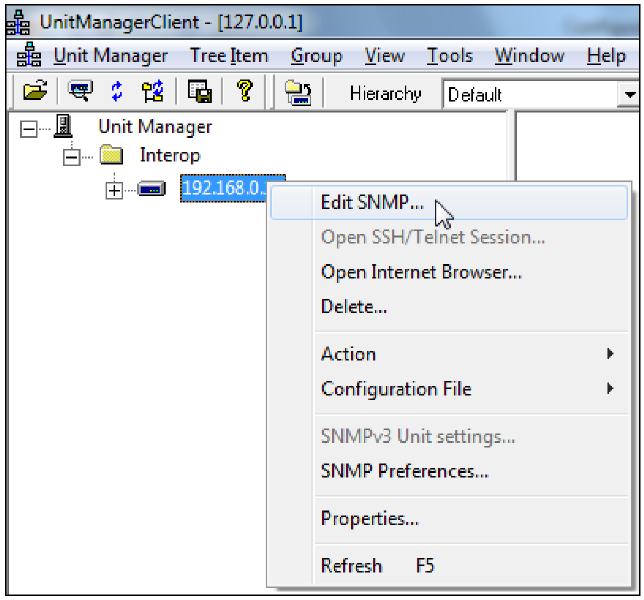

Figure 37

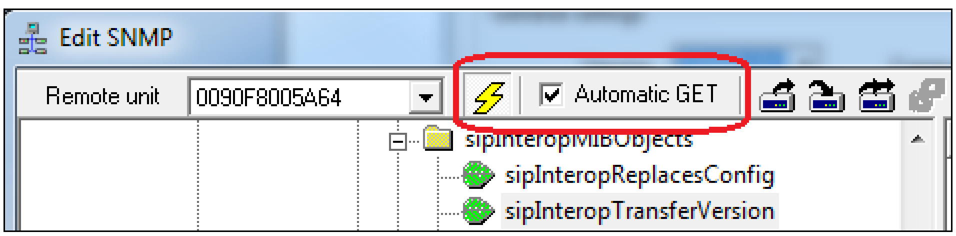

Figure 38

- Navigate to the desired variable by following the ISO tree, or simply by using the Find feature, which will be faster. Right click on any variable in the tree and click on "Find" as shown in figure 39 below:

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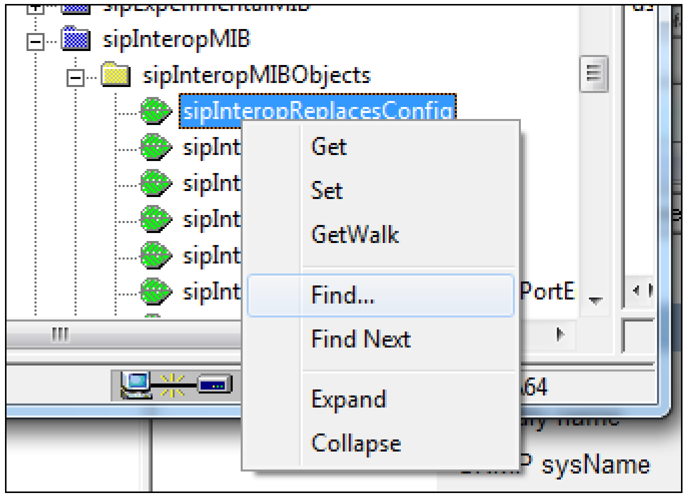

Figure 39

- Enter the Node Name as shown figure 40 below:

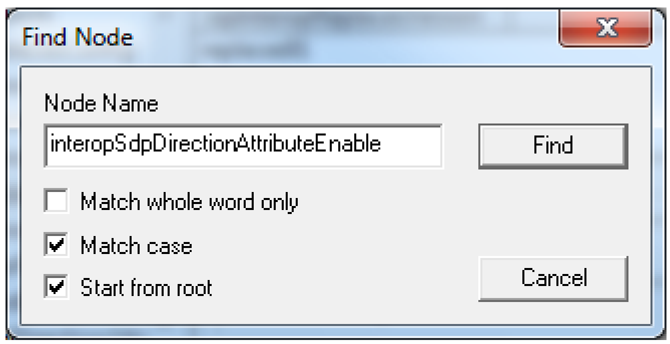

Figure 40

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5.6.6. Direction Attribute in the SDP

Disable the variable ***interopSdpDirectionAttributeEnable***

Navigate to the *interopSdpDirectionAttributeEnable* variable. The variable can be found by using the find feature or by navigating to the following iso -> org -> dod -> internet -> private -> mediatrix -> mediatrixSystem -> gen5 -> mediatrixCommon -> mediatrixServices -> sipEpMIB -> sipEpMIBObjects -> interopGroup (see figure 41)

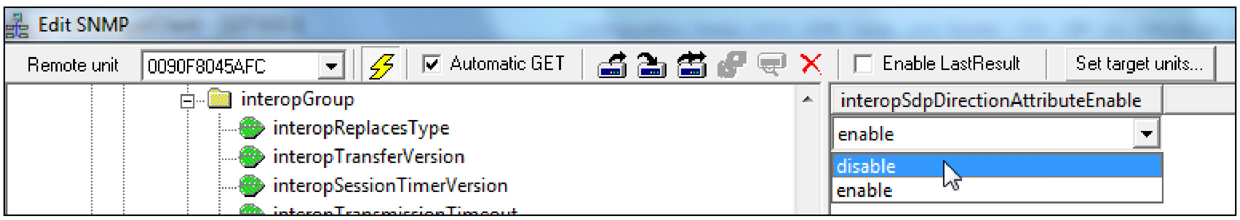

Figure 41

Click the **Set** button as figure 42 below:

Figure 42

+ Registration parameters adjustment

Set the defaultRegistrationProposedExpirationValue to **3600 (see figure 43)**

Navigate to the *defaultRegistrationProposedExpirationValue* variable (iso -> org -> dod -> internet -> private -> mediatrix -> mediatrixSystem -> gen5 -> mediatrixCommon -> mediatrixServices -> sipEpMIB -> sipEpMIBObjects -> registrationGroup)

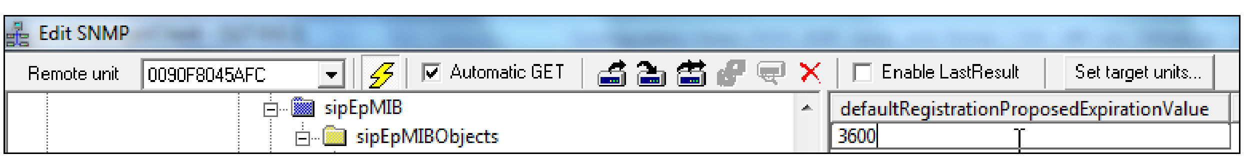

Figure 43

Click the Set button as mentioned on figure 42 above.

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When finished simply reboot the unit via the web page. You should now be able to use the Mediatrix 4100 unit with the Avaya CS1000.

6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the call establishment between the analog phones that connected to M4104 gateway and the CS1000 telephones; IP and SIP. Other call features, such as busy, hold, DTMF, MWI and codec negotiation, were exercised.

6.1. General test approach

The general test approach was to have one of the CS1000 telephone clients/users to place a call to and from the analog phone that connected to M4104 gateway and also to exercise other telephony features. The main objectives were to verify the M4104 successfully perform the following:

- Registration of analog phones which connected to M4104 port to the CS1000.
- Call establishment from analog phones with Avaya CS1000 SIP and non SIP phones/clients
- Call establishment from analog phones with emulated PSTN phones.
- Basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold, blind and supervised/consultative transfer, call waiting, second call.
- Advance CS 1000 Call Server features: speed dial, group call pickup, ring again busy/no answer, call park/retrieve, call forward (busy/all call/no answer), conference and multiple appearances DN
- Codec negotiations.

6.2. Test Results

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

- Avaya has not performed audio performance testing or reviewed the analog phones compliance to required industry standards.
- Enable Network Call Redirection (NCRD) in CS1000 Call Server SIP Line Route will cause an issue with Call Waiting. CR Q02073088 has been raised against CS 1000 SIP Line system. This has been no fixed plan so far.
- In the blind call transfer scenario, after completing transfer, BYE messages were sent by both SLG and transferor. This results in “481 Call Leg Does Not Exist” from one of the parties.
  The call flow is as follow:
  1. UA_1, dial Sigma Phone number

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2. Sigma Phone answers the call.
3. From Sigma Phone presses "HOLD" and then "TRANSFER" key to dial UA_2 number.
4. UA_ 2 answers the call.
5. UA_ 2 disconnects

This does not impact the feature operation.

- On the Call Park/Retrieve scenario, the call flow is as follow:
  1. From UA_01, dial UA_02
  2. From UA_02, park the call with the UA_01 by initiating a blind transfer and dial SPRE (7) + 71 followed by park to DN.
  3. From UA_02, retrieve the parked call by dialing SPRE (7)+72 followed by park to DN.

  In step 2 above, Flash key on UA_01 is pressed. A blind transfer is initiated by dialing SPRE (7) + 71 followed by park to DN. After pressing # key on UA_02 to complete the transfer, we expect that **UA_02 goes back to Idle status**. However, **UA_02 hears ring back tone continuously** until it is hanged up.
  This behavior does not impact the feature operation. The call is still re-connected normally when the 3rd party phone is un-parked. This is design intend of the Mediatrix 4104.

7. Verification Steps

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

- Verify that the analog phone registers successfully with the CS 1000 SIP Line Gateway server and Call Server by using CS 1000 Linux command line and CS 1000 Call Server overlay LD 32
  - Login sipline server using Avaya account.
  - Issue command "slgSetShowByUID [userID]" where userID is SIP Line user's ID being checked

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

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IP:Port:Trans = 47.248.100.56:5060:udp
         Type          = SIP3
         UserAgent     = PolycomVVX-VVX_1500-UA/3.2.2.0481
         x-nt-guid     = 9ad7a4871842718a35aeadf070608745
         RegDescrip    =
```


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```
RegStatus      = 1
PbxReason      = OK
SipCode        = 200
Expire         = 300
Contact        = sip:55524@47.248.100.56:5060
Nonce          = cad64489bbd9bca62aa9a1f833052da4
NonceCount     = 3
hTimer         = 0x9c659d0
TimeRemain     = 183
Stale          = 0
Outbound       = 0
ClientGUID     = 0

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

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

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

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

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

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are being investigated and solved by Mediatrix and Avaya design teams. Some of these issues are considered as exceptions. The M4104 gateway version Dgw 2.0.9.144 is considered compliant with CS1000 SIP Line System Release 6.0.

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9. Additional References

Product documentation for Avaya products may be found at:
http://support.nortel.com/go/main.jsp

[1] *Communication Server 1000 SIP Line Fundamental, Release 6.0, Revision 01.08, February 2010, Document Number NN43001-508*

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

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

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

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

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

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

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

Product information for Mediatrix 4104 products can be found at
http://www.media5corp.com

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10. Appendix

This section is to help user provisioning the clock synchronization of a network of CS1000 systems used under test. This will create the PRI trunk synchronization between 2 CS1000 under test in Figure 1; Main and Emulated PSTN systems. In this example, the emulated PSTN will have clock controller card. Therefore, these provisioning steps below will only apply to the emulated PSTN CS1000 system only. The steps below can be accomplished by login Call Server and using command line interface on overlay 73.

Defining system timers and clock controller parameters

The programming example below shows how to define system timers and clock controller parameters using LD 73. For all other fields not listed in the example press RETURN to use default values.

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

Enabling T1 QSIG Service

Enable TMDI card

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

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

OK
```

Enable Clock Controller

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

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


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Enable PRI loop

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

```
>ld 60
DTI000
.enll 10

OK
.
```

Enable D-Channel

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

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

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

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


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