



## **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.

# 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.

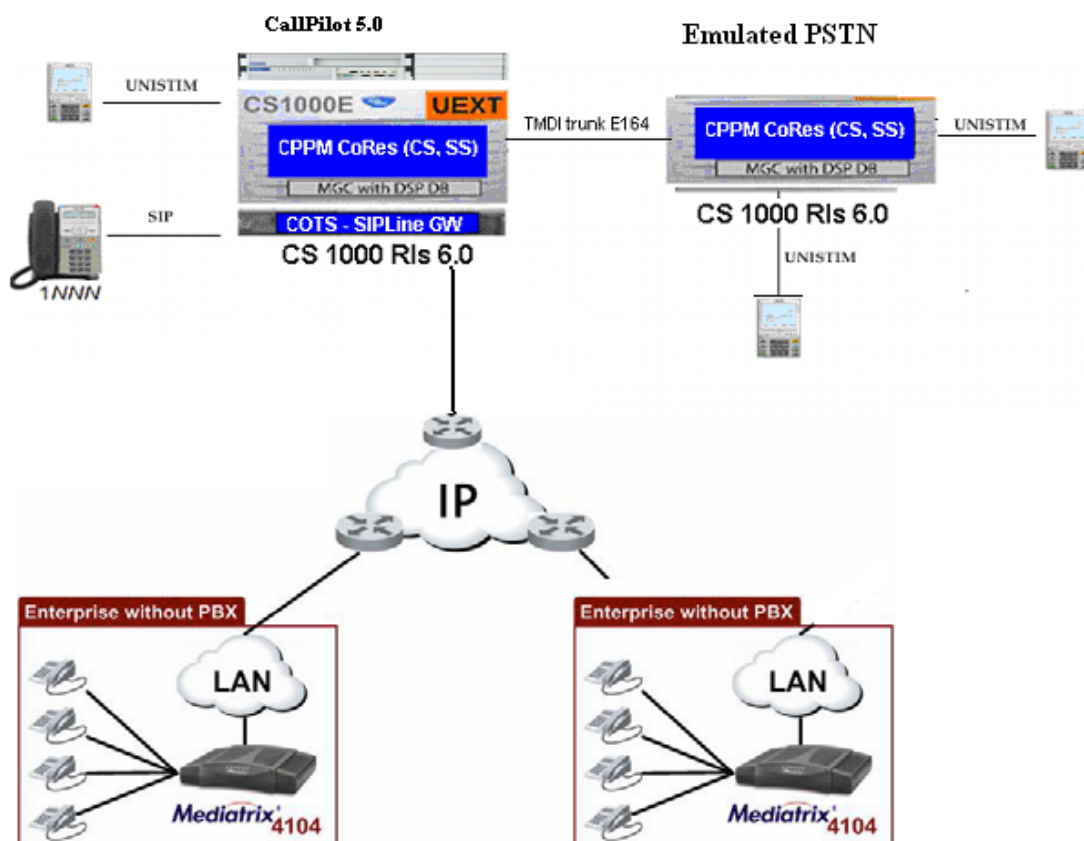


Figure 1

### 3. Equipment and Software Validated

System	Software Version
CS1000	<ul style="list-style-type: none"> <li>Call Server (CPPM): 6.00RJ</li> <li>Signalling Server (CPPM): 6.00.18</li> <li>SIP Line Gateway (HP DL320)</li> </ul>
Voicemail system	<ul style="list-style-type: none"> <li>CallPilot 5.0 system</li> </ul>
11xx SIP client (Sigma)	<ul style="list-style-type: none"> <li>02.02.16.00</li> </ul>
SIP soft-phones	<ul style="list-style-type: none"> <li>SMC3456: v2.6 Build 53715</li> </ul>
IP phones	<ul style="list-style-type: none"> <li>2050PC: 3.02.0045</li> </ul>
Analog phone	<ul style="list-style-type: none"> <li>Northern Telecom, NTD 9519</li> </ul>
Mediatix 4104	<ul style="list-style-type: none"> <li>Dgw 2.0.9.144</li> </ul>

## 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
  - o Installed with CS 1000 Release 6.0 Linux Base.
  - o Joined CS 1000 Release 6.0 Security Domain.
  - o Deployed with SIP Line Application.

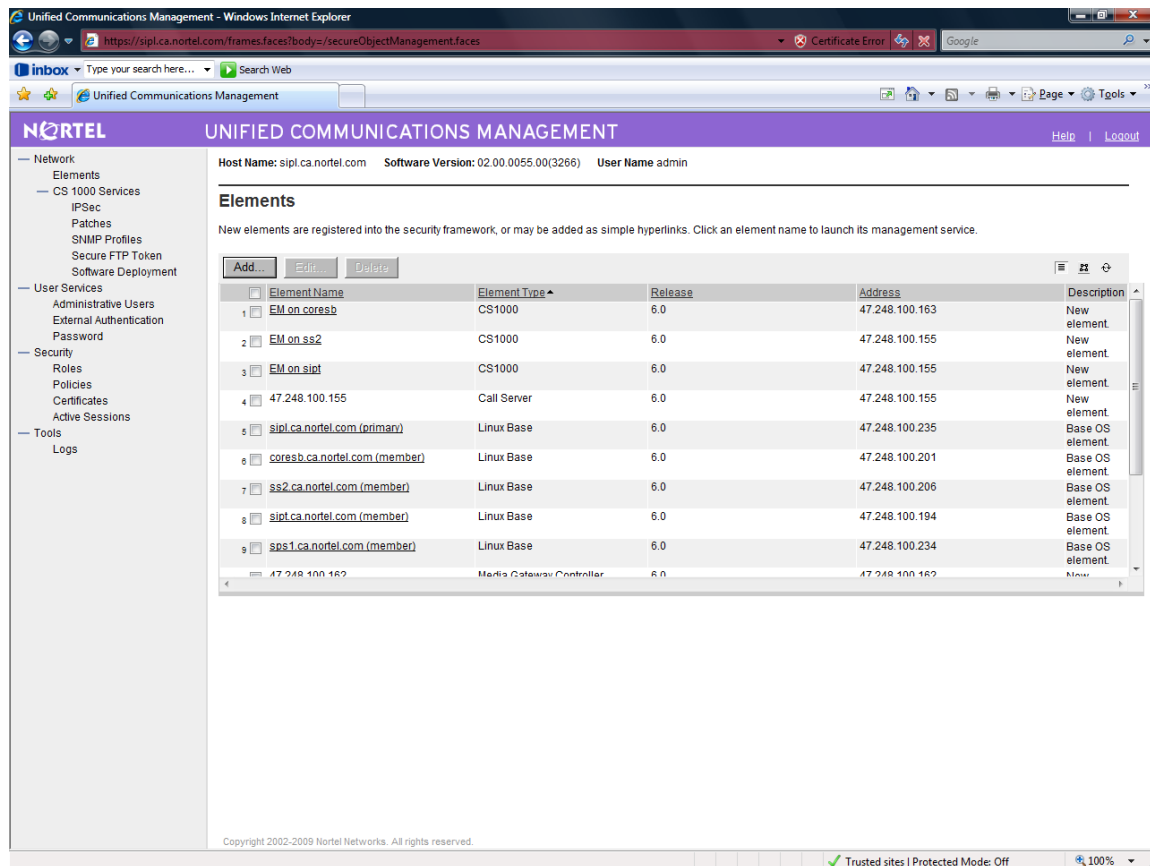
For more information, see [6].

- Following packages are enabled in the keycode.
- If it has not been enable, please contact Avaya technical support at [www.avaya.com](http://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].



**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.

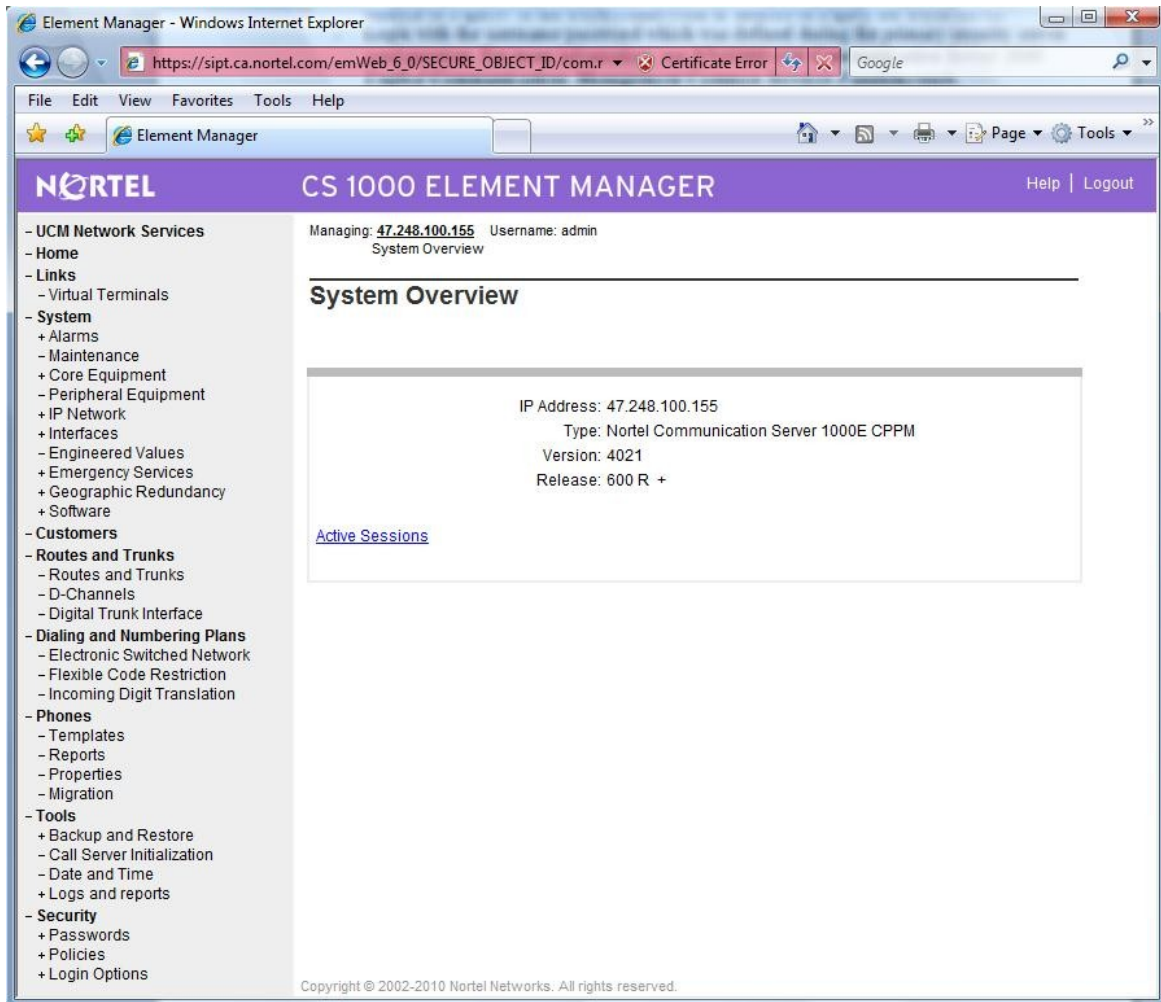
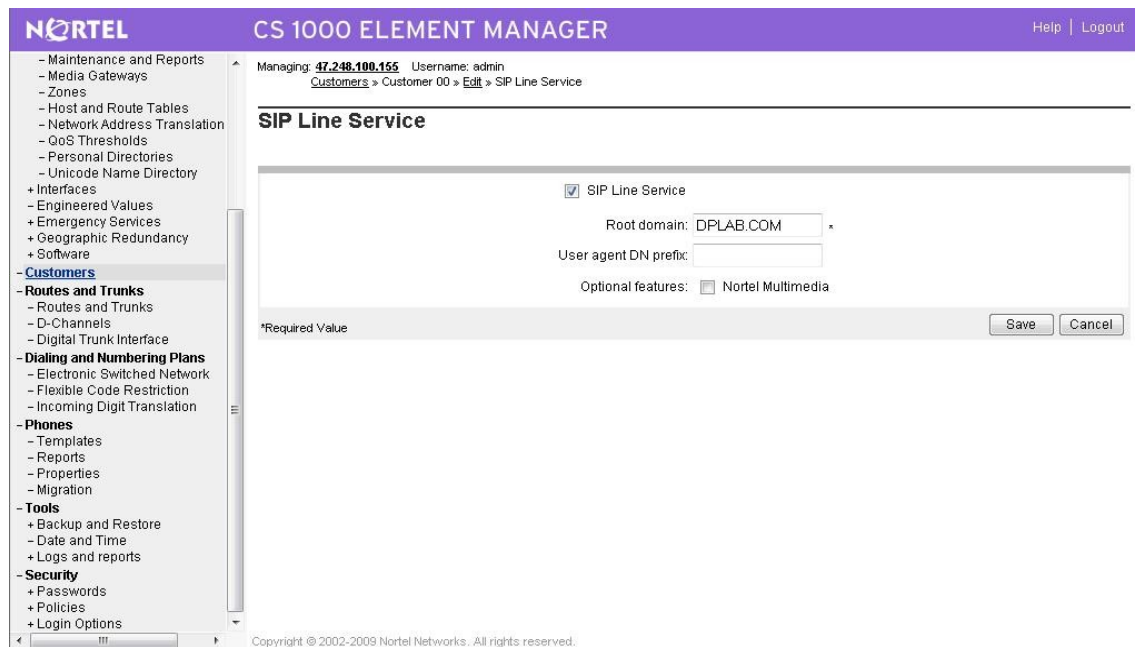


Figure 3: CS 1000 EM Home Page

#### 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.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System » IP Network » IP Telephony Nodes

### New IP Telephony Node

Step 1: Define the new Node and its services.  
You will also require pre-configured servers with appropriate application software already deployed to host the selected services.

Node ID: 556 \* (0-9999)

Call Server IP Address: 47.248.100.155 \*

**Telephony LAN (TLAN)**  
Node IP Address: 47.248.100.237 \*  
Subnet Mask: 255.255.255.240 \*

**Embedded LAN (ELAN)**  
Gateway IP address: 47.248.100.129 \*  
Subnet Mask: 255.255.255.224 \*

Applications ☒ SIP Line  
☐ UNISim Line Terminal Proxy Server (LTPS)  
☐ Virtual Trunk Gateway (SIPGw, H323Gw)  
☐ Personal Directory (PD)  
☐ Presence Publisher

\* Required Value. Next > Cancel

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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**.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System » IP Network » IP Telephony Nodes

### New IP Telephony Node (ID:557)

Step 2: Associate required signaling servers for SIP Line services.  
In order to appear in the list below, servers must already be defined within ECM, should not be part of any other IP telephony node and deployed application(s) on the server(s) should match the service(s) selected for this node.

sip Add Remove Make Leader Print Refresh

<input type="checkbox"/>	Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
Select from the list above and click Add to associate servers with this node. Selected servers must have identical application deployments.						

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**Figure 6 – IP Telephony Node – Add Server**

- 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**.

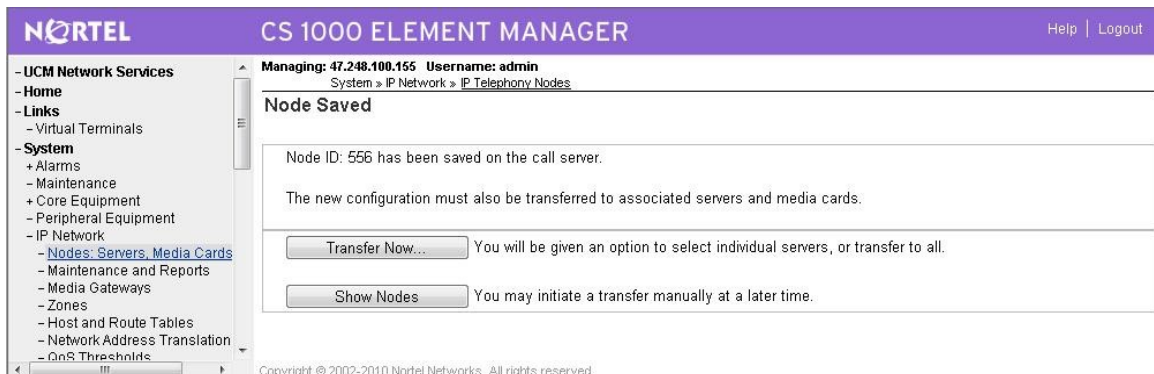
**Figure 7 – SIP Line Node Details**

- 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.)**

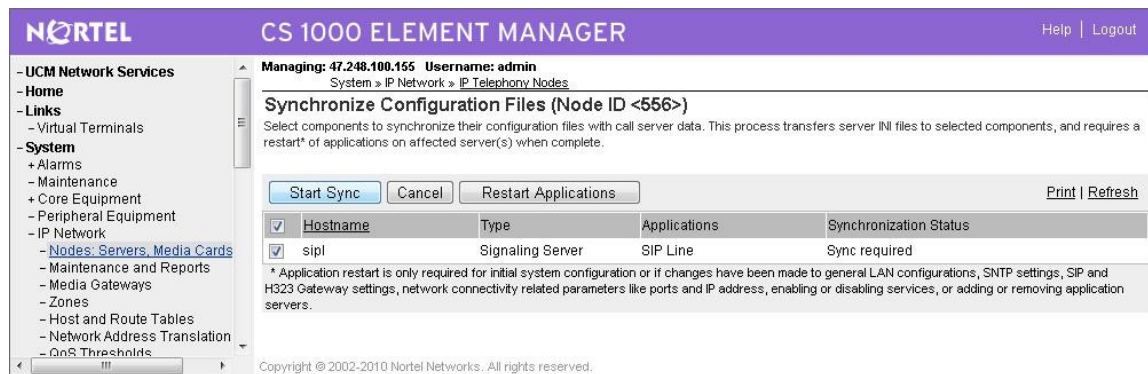
- Click **Next**. The **Confirm new Node details** page appears (not shown).

- 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)**.

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

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin  
Routes and Trunks > D-Channels > D-Channels 30 Property Configuration

### D-Channels 30 Property Configuration

**- Basic Configuration**

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	SIPLine
Recovery to Primary (RCVP)	<input type="checkbox"/>
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
Country (CNTY)	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> more PRI
Secondary PRI2 loops (PRI2)	<input type="text"/>
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	5
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000
<b>- Basic options (BSCOPT)</b>	
Primary D-channel for a backup DCH (PDCH)	Range: 0 - 254
- PINX customer number (PINX_CUST)	
- Progress signal (PROG)	
- Calling Line Identification (CLID)	
- Output request Buffers (OTBF)	32
- D-channel transmission Rate (DRAT)	56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option (CNEG)	No alternative acceptable, exclusive. (1)
- Remote Capabilities (RCAP)	<a href="#">Edit</a>
<b>+ - Change protocol timer value (TIMR)</b>	
- B channel Service messaging. (BSRV)	<input type="checkbox"/>
<b>+ Advanced options (ADVOPT)</b>	
<b>+ Feature Packages</b>	

Submit Refresh Delete Cancel

**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.

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin  
Routes and Trunks » D-Channels » D-Channels 30 Property Configuration » - Remote Capabilities Configuration

### - Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3)	<input type="checkbox"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDL card (MSL)	<input type="checkbox"/>
Message waiting interworking with DMS-100 (MWI)	<input checked="" type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
Network name display method 2 (ND2)	<input checked="" type="checkbox"/>
Network name display method 3 (ND3)	<input type="checkbox"/>
Name display - integer ID coding (NDI)	<input type="checkbox"/>
Name display - object ID coding (NDO)	<input type="checkbox"/>
Path replacement uses integer values (PRI)	<input type="checkbox"/>
Path replacement uses object identifier (PRO)	<input type="checkbox"/>
Release Link Trunks over IP (RLTI)	<input type="checkbox"/>
Remote virtual queuing (RVQ)	<input type="checkbox"/>
Trunk anti-froboning operation (TAT)	<input type="checkbox"/>
User to user service 1 (UUS1)	<input type="checkbox"/>
NI-2 name display option. (NDS)	<input type="checkbox"/>
Message waiting indication using integer values (QMWI)	<input type="checkbox"/>
Message waiting indication using object identifier (QMWIO)	<input type="checkbox"/>
User to user signalling (UUI)	<input type="checkbox"/>

Return - Remote Capabilities    Cancel

**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.

## 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.

The screenshot shows the 'New Application Module Link' configuration page in the Nortel CS 1000 Element Manager. The page has a purple header with the Nortel logo and 'CS 1000 ELEMENT MANAGER'. A navigation tree on the left includes 'UCM Network Services', 'Home', 'Links', 'System', 'Interfaces', 'Application Module Link', 'Value Added Server', 'Property Management System', 'Engineered Values', 'Emergency Services', 'Geographic Redundancy', 'Software', 'Customers', and 'Routes and Trunks'. The main content area shows the following fields: 'Port number' set to 32 (range 16-127), 'AML over ELAN' checked, 'Description' set to 'SIPLine', 'Link control system parameters' unchecked, and 'Maximum octets' set to 512 (per HDLC frame). 'Save' and 'Cancel' buttons are at the bottom right. The footer contains the copyright notice: 'Copyright © 2002-2010 Nortel Networks. All rights reserved.'

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.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System > Interfaces > Value Added Server > Edit Value Added Server 032

### Edit Value Added Server 032

Ethernet LAN Link: 032  
ELAN port configured in ADAN

Application Security:

Interval: 1  
Time interval for checking the link for overload in five second increments

Message Count Threshold: 9999 \* (10 - 9999)

Save Cancel

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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.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System > IP Network > Zones > Bandwidth Zones > Bandwidth Zones 254 > Zone Basic Property and Bandwidth Management

### Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	254
Intrazone Bandwidth (INTRA_BW):	100000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	100000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

Submit Refresh Delete Cancel

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**Figure 15 – Virtual Trunk Zone Configuration**

## 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**.

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Managing: 47.248.100.155 Username: admin  
Routes and Trunks > Routes and Trunks > Customer 0, Route 30 Property Configuration

### Customer 0, Route 30 Property Configuration

**- Basic Configuration**

Route data block (RDB) (TYPE)

Customer number (CUST)

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

Trunk type M911P (M911P) ☐

The route is for a virtual trunk route (VTRK) ☒

Zone for codec selection and bandwidth management (ZONE)  Range: 0 - 255

Node ID of signaling server of this route (NODE)  Range: 0 - 9999

Protocol ID for the route (PCID)

Integrated services digital network option (ISDN) ☒

Mode of operation (MODE)

D channel number (DCH)  Range: 0 - 254

Interface type for route (IFC)

Private network identifier (PNI)  Range: 0 - 32700

Network calling name allowed (NCNA) ☒

Network call redirection (NCRD) ☒

Trunk route optimization (TRO) ☐

Recognition of DTI2 ABCD FALT signal for ISL (FALT) ☐

Channel type (CHTY)

Call type for outgoing direct dialed TIE route (CTYP)

Insert ESN access code (INAC) ☒

Integrated service access route (ISAR) ☐

Display of access prefix on CLID (DAPC) ☐

Mobile extension route (MBXR) ☐

**+ Basic Route Options**

**+ Network Options**

**+ General Options**

**+ Advanced Configurations**

Submit Refresh Delete Cancel

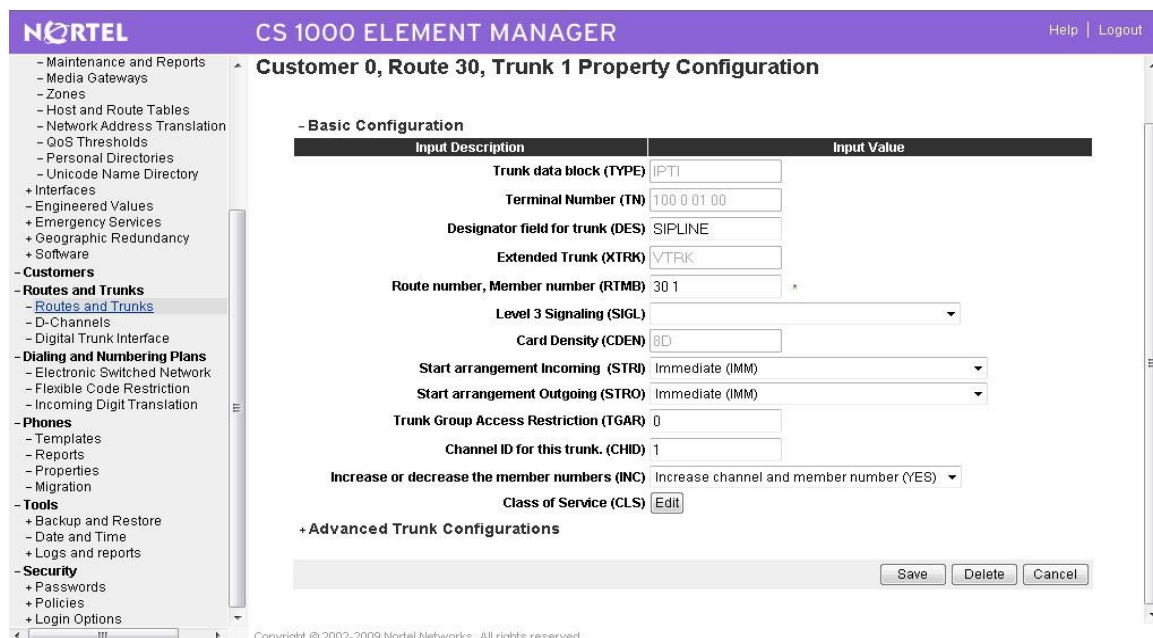
Figure 16 –SIP Line Route Configuration

**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.

- 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**.



**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

**Customer 0, Route 30, Trunk 1 Property Configuration**

**- Basic Configuration**

Input Description	Input Value
Trunk data block (TYPE)	IPT1
Terminal Number (TN)	100 0 01 00
Designator field for trunk (DES)	SIPLINE
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	30 1
Level 3 Signaling (SIGL)	
Card Density (CDEN)	8D
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk. (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	<a href="#">Edit</a>

**+ Advanced Trunk Configurations**

Save Delete Cancel

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**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

**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 RCBF  
 ICDD CDMD LLCN MCTD CLBD AUTU  
 GPU A DPU A **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 CCBF 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

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 MSDI 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
--------	----------	-------------

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.

		<b>Note:</b> 2.0 Mb only supports PRI or SHA user
IFC	ISGF	Interface type.
DCHL	10	<p>PRI card number carries the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above.</p> <p>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.</p> <p>xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.</p>
SIDE	NET	<p>NET = network, the controlling switch (applied for CS 1000 PSTN simulator)</p> <p>USR = slave to the controller (applied for CS 1000 system under test)</p>
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	DIV 2 3	Path Replacement Triggers
PR_TRIGS	CNG 2 3	
PR_TRIGS	CON 2 3	
PR_TRIGS	CTR2 2 3	

#### 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.

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.

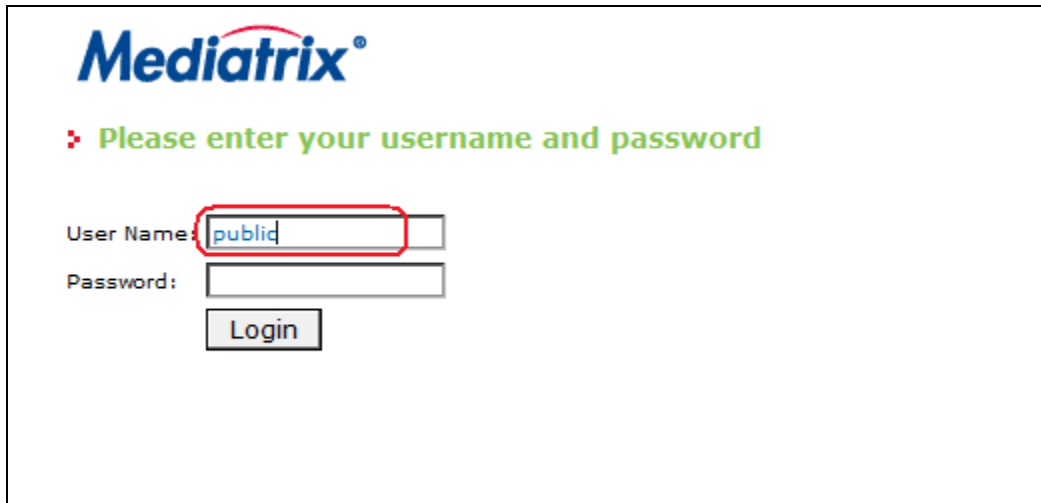
Prompt	Response	Description
REQ	NEW 23	Create 23 new trunks
TYPE	Tie	Trunk type
TN	10 1	Loop (card) and channel number for digital trunks
PCML	MU	System PCM law.
DES	T1_QSIG	Designator field for trunk
CUST	0	Customer number
RTMB	10 1	Service route number and trunk member number
CLS	UNR DTN	Trunk Class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



**Mediatrix®**

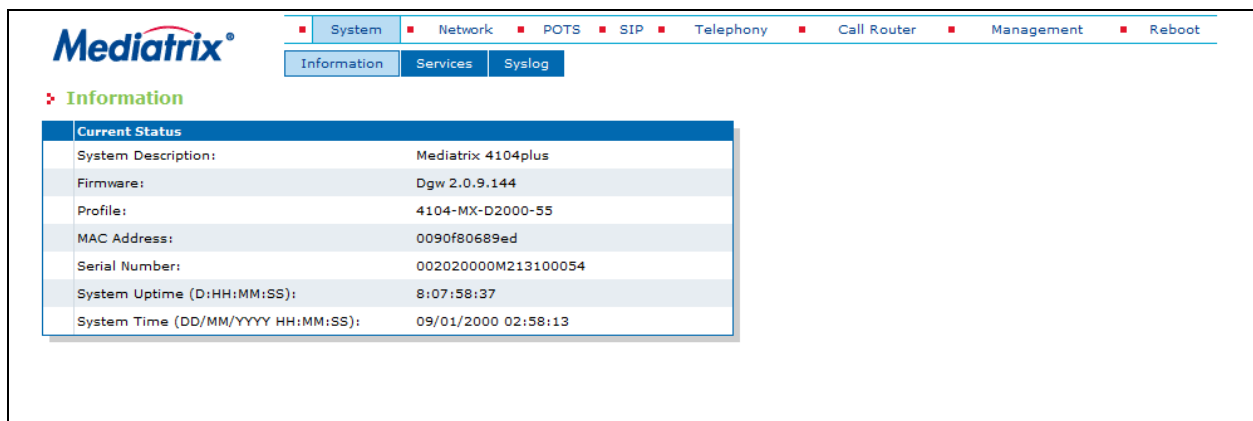
❖ Please enter your username and password

User Name:

Password:

**Figure 18 – Login Screen**

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



**Mediatrix®**

System Network POTS SIP Telephony Call Router Management Reboot

Information Services Syslog

❖ Information

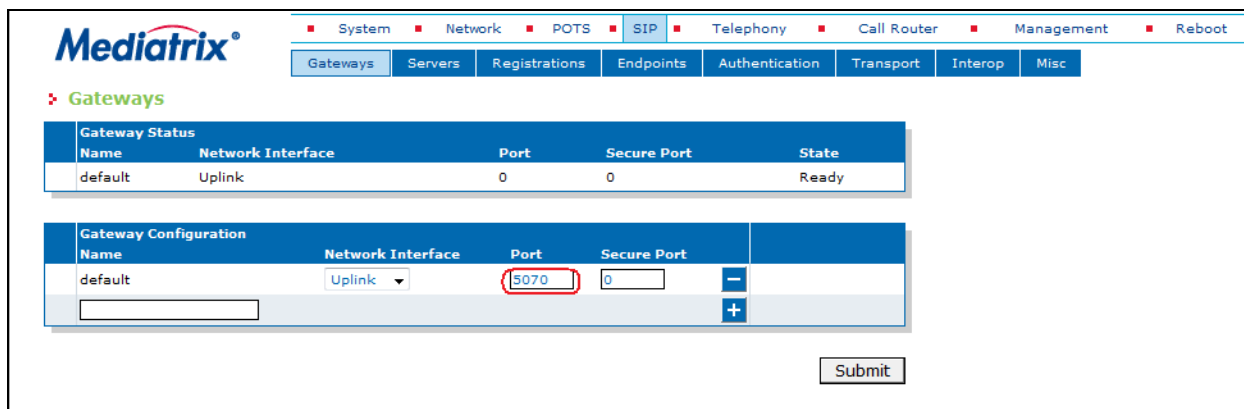
Current Status	
System Description:	Mediatrix 4104plus
Firmware:	Dgw 2.0.9.144
Profile:	4104-MX-D2000-55
MAC Address:	0090f80689ed
Serial Number:	002020000M213100054
System Uptime (D:HH:MM:SS):	8:07:58:37
System Time (DD/MM/YYYY HH:MM:SS):	09/01/2000 02:58:13

**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.



The screenshot shows the Mediatrix web interface with the 'SIP' tab selected. The 'Gateways' section is expanded, showing a table with gateway status and a configuration table below it. The configuration table has a 'Port' field set to 5070, which is circled in red. A 'Submit' button is at the bottom right.

Gateway Status				
Name	Network Interface	Port	Secure Port	State
default	Uplink	0	0	Ready

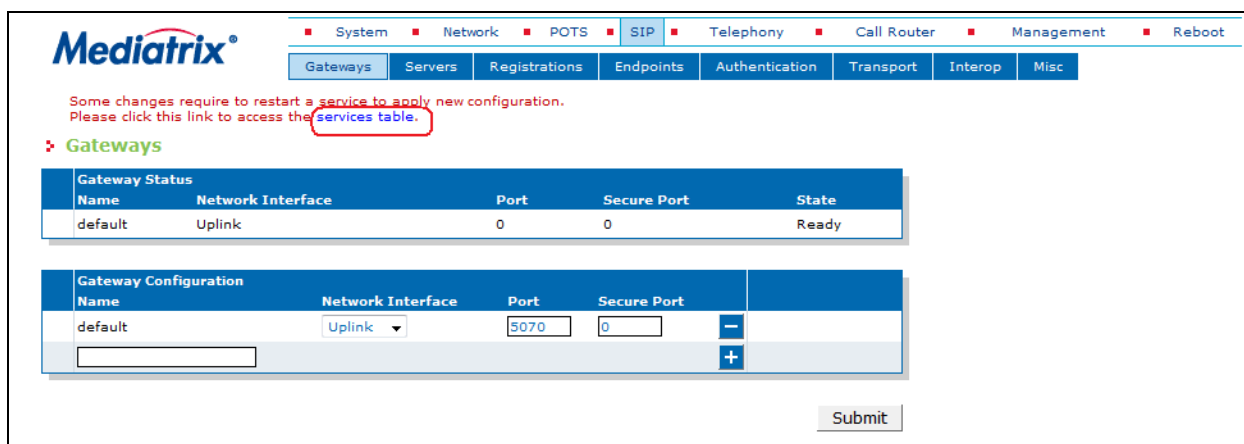
  

Gateway Configuration				
Name	Network Interface	Port	Secure Port	
default	Uplink	5070	0	-
				+

Submit

**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.



The screenshot shows the Mediatrix web interface with a red message: 'Some changes require to restart a service to apply new configuration. Please click this link to access the services table.' The 'services table' link is circled in red. The 'Gateways' section is expanded, showing the same gateway status and configuration tables as in Figure 20. A 'Submit' button is at the bottom right.

Some changes require to restart a service to apply new configuration.  
Please click this link to access the [services table](#).

Gateways

Gateway Status				
Name	Network Interface	Port	Secure Port	State
default	Uplink	0	0	Ready

Gateway Configuration				
Name	Network Interface	Port	Secure Port	
default	Uplink	5070	0	-
				+

















Submit

**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.

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	  	
<b>SIP Endpoint (SIEP):</b>	<b>Started</b>	<b>Auto</b>	  	<b>Restart needed</b>
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.

Successfully sent the services restart command. Click [here](#) to get the latest statuses.

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

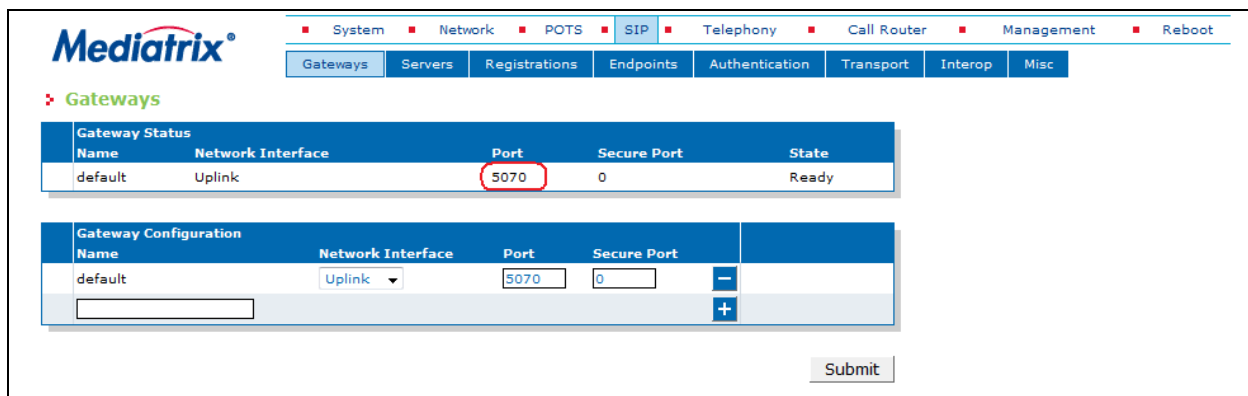
User Service	Status	Startup Type	Action	Comment
Basic Network Interface (BNI):	Started	Auto	  	
Call Routing (CROUT):	Started	Auto	  	
Command Line Interface (CLI):	Started	Auto	  	
DHCP Server (DHCP):	Stopped	Manual	  	
Endpoint Administration (EPADM):	Started	Auto	  	
Endpoint Services (EPSERV):	Started	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	  	
Simple Network Management Protocol (SNMP):	Started	Auto	  	
Telephony Interface (TELIF):	Started	Auto	  	
Web (WEB):	Started	Auto	  	

Submit

Restart Required Services

Figure 23

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



The screenshot shows the Mediatrix configuration interface with the 'SIP' tab selected. Under the 'Gateways' section, there are two tables. The first table, 'Gateway Status', shows a single entry for 'default' with 'Uplink' as the network interface, '5070' as the port (highlighted with a red circle), '0' as the secure port, and 'Ready' as the state. The second table, 'Gateway Configuration', shows the same 'default' gateway with 'Uplink' as the network interface, '5070' as the port, and '0' as the secure port. There are minus and plus buttons next to the secure port field. A 'Submit' button is at the bottom right.

Gateway Status				
Name	Network Interface	Port	Secure Port	State
default	Uplink	5070	0	Ready

Gateway Configuration				
Name	Network Interface	Port	Secure Port	
default	Uplink	5070	0	-
				+

Submit

**Figure 24: Confirming the SIP Port**

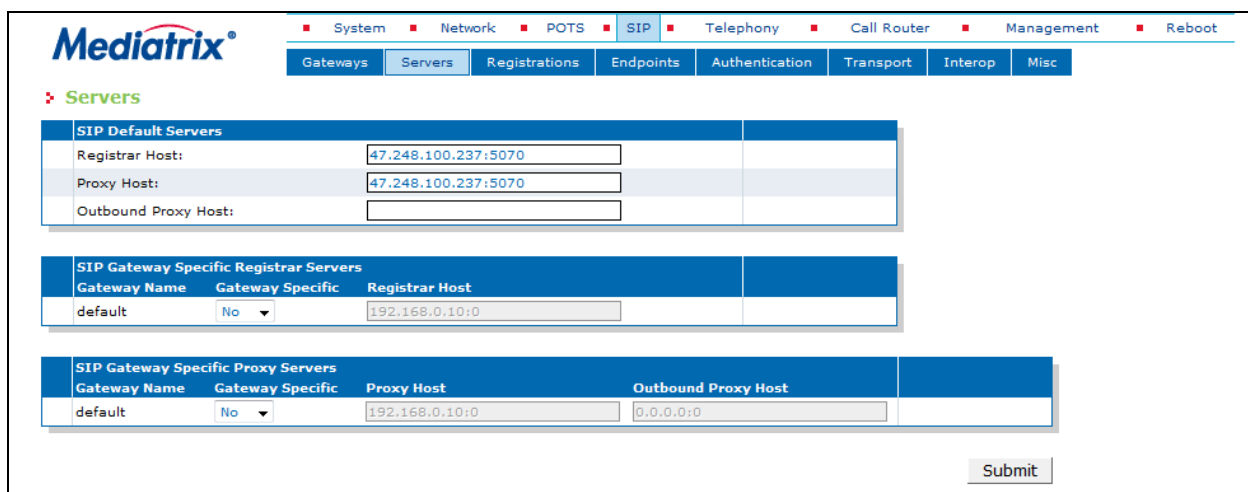
### 5.3. Configure communication to the SIP Proxy Server

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.



The screenshot shows the Mediatrix configuration interface with the 'SIP' tab selected and the 'Servers' sub-tab active. There are three main sections: 'SIP Default Servers', 'SIP Gateway Specific Registrar Servers', and 'SIP Gateway Specific Proxy Servers'. Each section has a table with configuration fields. The 'SIP Default Servers' table has fields for Registrar Host, Proxy Host, and Outbound Proxy Host, all with values entered. The 'SIP Gateway Specific Registrar Servers' table has columns for Gateway Name, Gateway Specific, and Registrar Host. The 'SIP Gateway Specific Proxy Servers' table has columns for Gateway Name, Gateway Specific, Proxy Host, and Outbound Proxy Host. A 'Submit' button is at the bottom right.

SIP Default Servers		
Registrar Host:	47.248.100.237:5070	
Proxy Host:	47.248.100.237:5070	
Outbound Proxy Host:		

SIP Gateway Specific Registrar Servers		
Gateway Name	Gateway Specific	Registrar Host
default	No	192.168.0.10:0

SIP Gateway Specific Proxy Servers			
Gateway Name	Gateway Specific	Proxy Host	Outbound Proxy Host
default	No	192.168.0.10:0	0.0.0.0:0

Submit

**Figure 25: Servers Screen**

## 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.

Endpoints Registration Status				
Endpoint	User Name	Gateway Name	Registrar	Status
Port1	55111	default	47.248.100.237:5070	Registered
Port2	55112	default	47.248.100.237:5070	Registered
Port3	55115	default	47.248.100.237:5070	Registered

Unit Registration Status			
User Name	Gateway Name	Registrar	Status

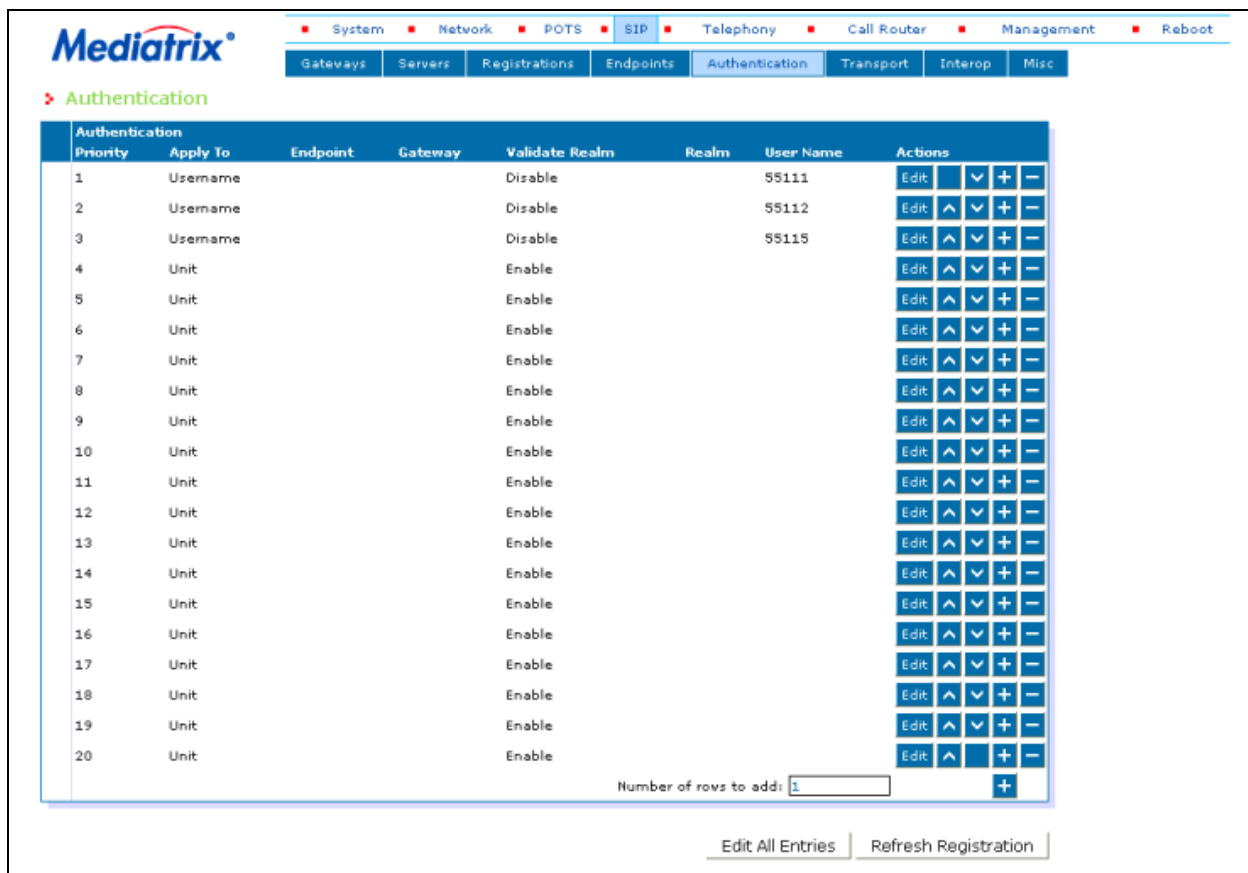
Endpoints Registration				
Endpoint	User Name	Friendly Name	Register	Gateway Name
Port1	55111		Enable	all
Port2	55112		Enable	all
Port3	55115		Enable	all
Port4			Disable	all

Unit Registration		
Index	User Name	Gateway Name

Submit Submit & Refresh Registration

Figure 26: Registrations Screen

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



**Figure 27: Authentications Screen**

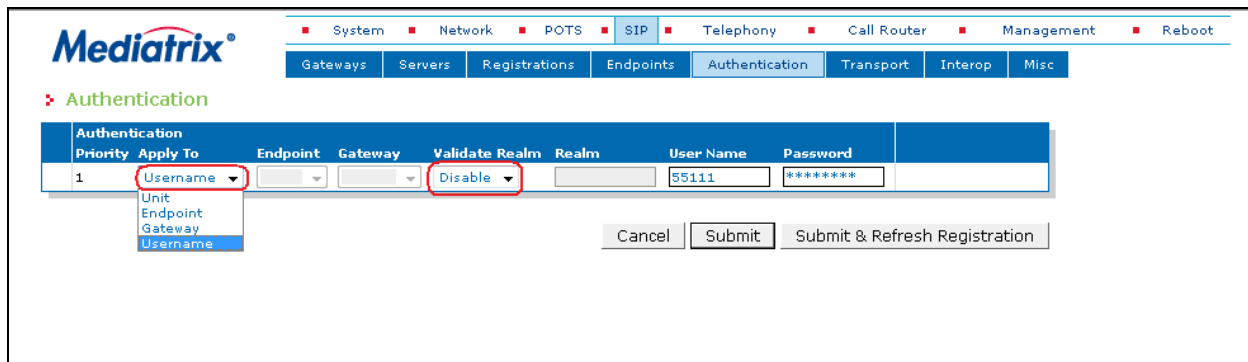
On the Authentication page, click on the Edit button to configure authentication method for each individual User as shown in figure 28.

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.

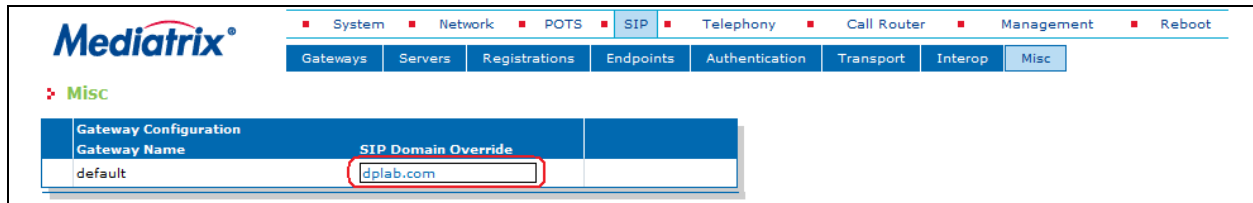


Figure 29

## 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.

## CODECS

Select Endpoint: Default

CODEC	Voice	Data	Advanced
G.711 a-Law	Enable	Disable	Edit
G.711 u-Law	Enable	Disable	Edit
G.723	Enable		Edit
G.726 16Kbps	Disable		Edit
G.726 24Kbps	Disable		Edit
G.726 32Kbps	Disable	Disable	Edit
G.726 40Kbps	Disable	Disable	Edit
G.729	Enable		Edit
T.38		Enable	Edit
Clear Mode	Disable	Disable	Edit
Clear Channel	Disable	Disable	Edit
X CCD	Disable	Disable	Edit

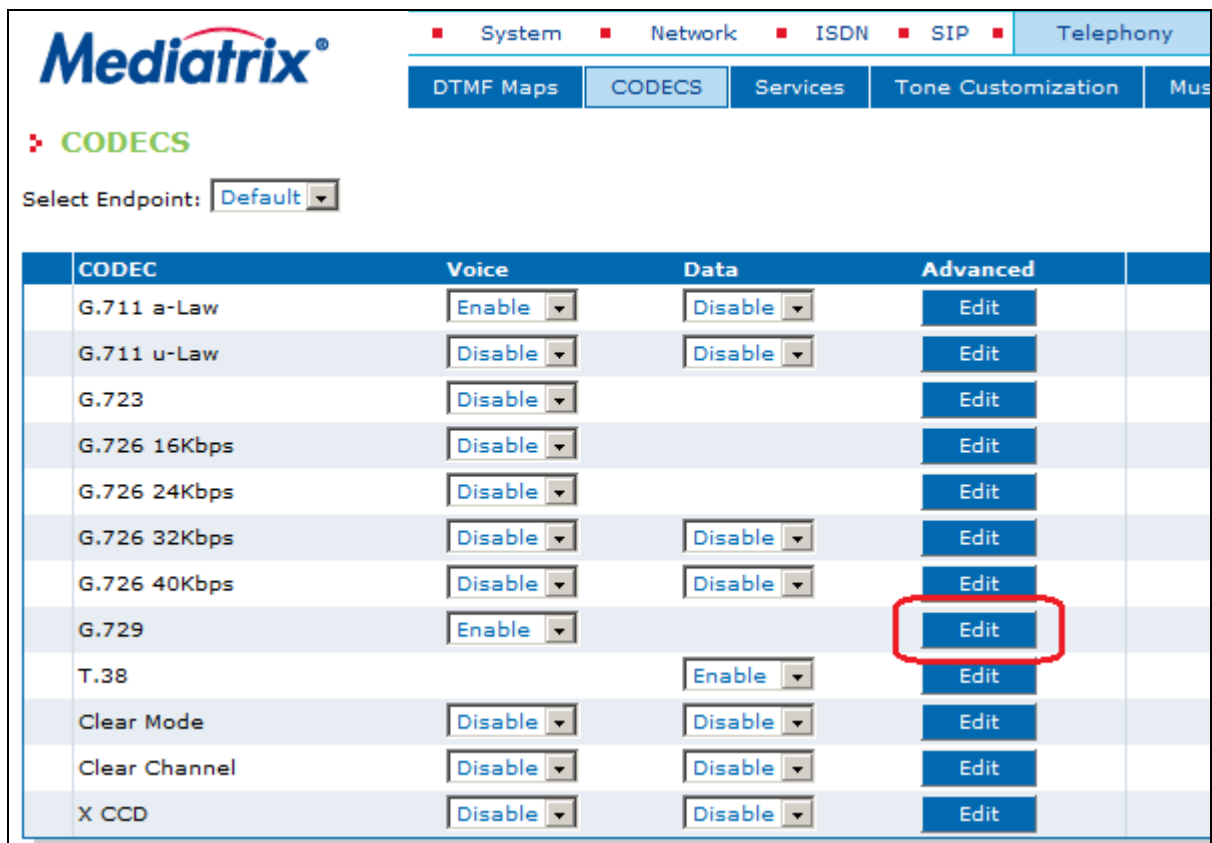
<b>Misc</b>	
<b>Jitter Buffer</b>	
Level:	Normal
Minimum:	30
Maximum:	125
<b>DTMF Transport</b>	
Transport Method:	Out-of-Band using RTP
SIP Transport Method:	Info DTMF Relay
Payload Type:	101
<b>Generic Voice Activity Detection (VAD)</b>	
Enable (G.711 and G.726):	Disable

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.

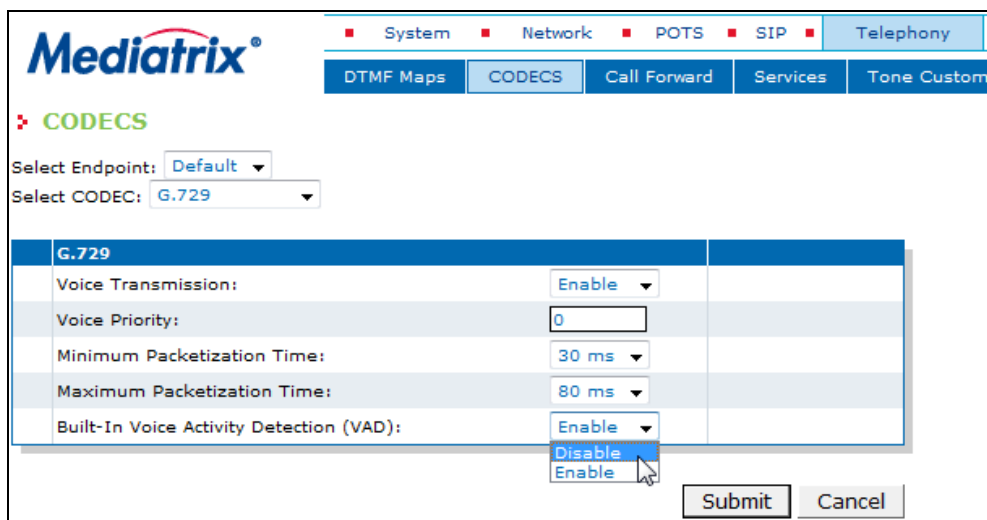


The screenshot shows the Mediatrix web interface for configuring CODECS. The 'Telephony' tab is selected, and the 'CODECS' sub-tab is active. A dropdown menu shows 'Default' as the selected endpoint. Below this is a table with columns for CODEC, Voice, Data, and Advanced settings. The 'G.729' row is highlighted, and its 'Edit' button is circled in red.

CODEC	Voice	Data	Advanced
G.711 a-Law	Enable	Disable	Edit
G.711 u-Law	Disable	Disable	Edit
G.723	Disable		Edit
G.726 16Kbps	Disable		Edit
G.726 24Kbps	Disable		Edit
G.726 32Kbps	Disable	Disable	Edit
G.726 40Kbps	Disable	Disable	Edit
G.729	Enable		Edit
T.38		Enable	Edit
Clear Mode	Disable	Disable	Edit
Clear Channel	Disable	Disable	Edit
X CCD	Disable	Disable	Edit

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.



The screenshot shows the Mediatrix web interface for configuring the G.729 CODEC. The 'Telephony' tab is selected, and the 'CODECS' sub-tab is active. The 'G.729' CODEC is selected in the dropdown menu. The 'Built-In Voice Activity Detection (VAD)' dropdown menu is open, showing 'Disable' as the selected option. The 'Submit' button is visible at the bottom right.

G.729	
Voice Transmission:	Enable
Voice Priority:	0
Minimum Packetization Time:	30 ms
Maximum Packetization Time:	80 ms
Built-In Voice Activity Detection (VAD):	Disable

Figure 32

### 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.

Misc	
<b>Jitter Buffer</b>	
Level:	Normal
Minimum:	30
Maximum:	125
<b>DTMF Transport</b>	
Transport Method:	Out-of-Band using RTP
Payload Type:	101
<b>Generic Voice Activity Detection (VAD)</b>	
Enable (G.711 and G.726):	Conservative
<b>Base Ports</b>	
RTP:	5004
T.38:	6004
SRTP:	5004

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.

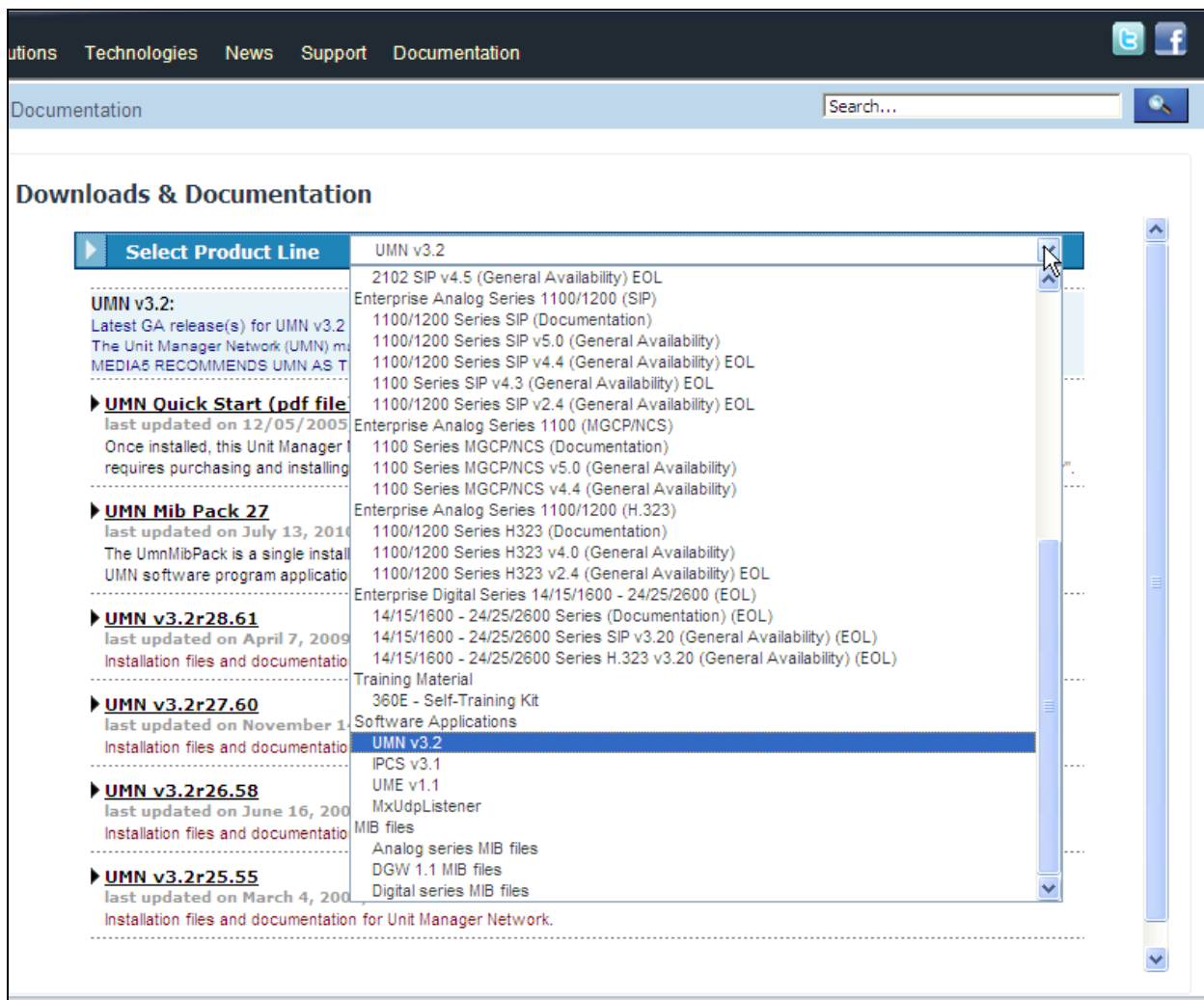


Figure 34

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.

### 5.6.2. Start the UMN

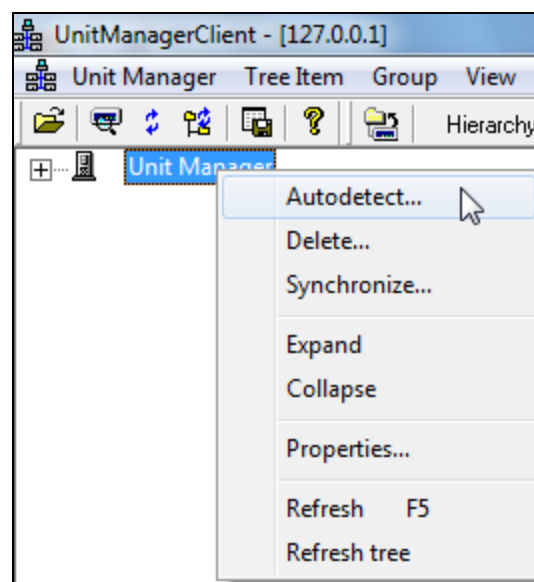
Select from the Start Menu > Programs > Unit Manager Network 3.2 > Unit Manager Network.

### 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.

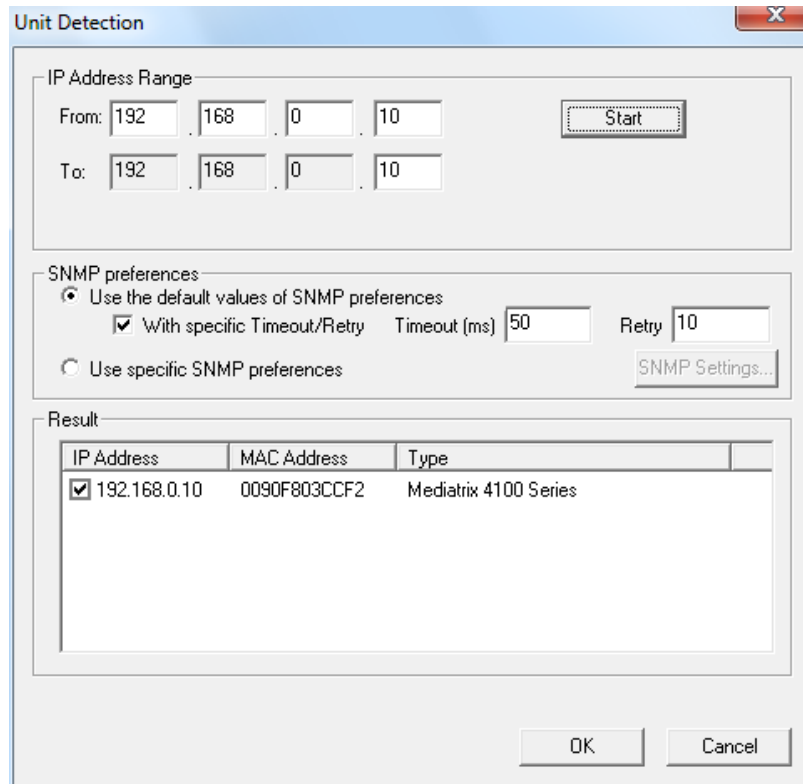
### 5.6.4. Perform an Autodetect

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.

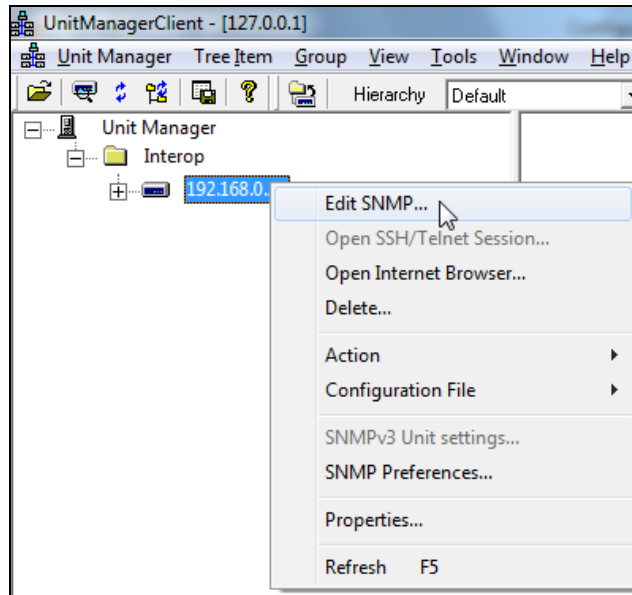


**Figure 36**

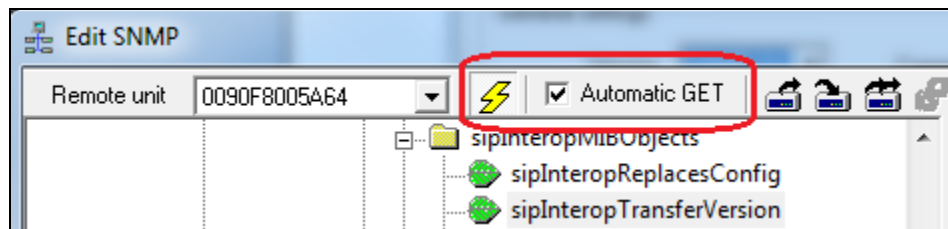
### 5.6.5. Advanced configuration

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.

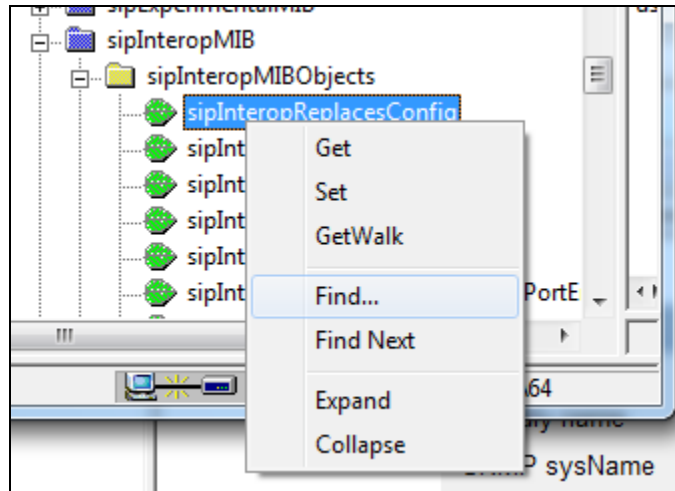


**Figure 37**



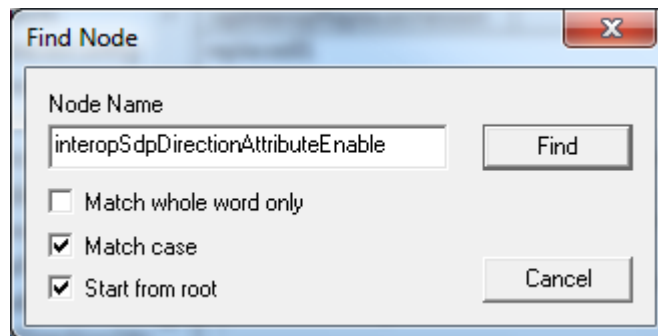
**Figure 38**

- 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:



**Figure 39**

- Enter the Node Name as shown figure 40 below:



**Figure 40**

### 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)

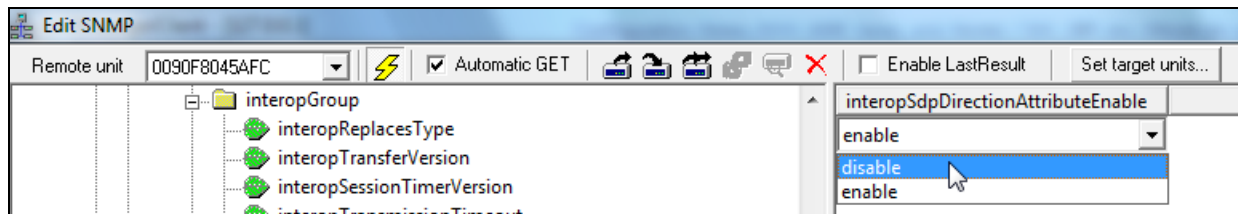


Figure 41

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

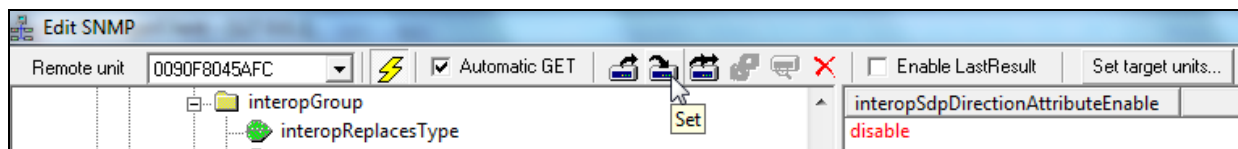


Figure 42

#### + 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.

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

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     = PolycomV VX-VVX_1500-UA/3.2.2.0481
x-nt-guid     = 9ad7a4871842718a35aeadf070608745
RegDescrip    =
```

```

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.

## 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>

## 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
```

### **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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