

Mediatrix Gateway 440x Series

Quick Configuration Guide

All BRI Mediatrix gateways are pre-configured on ETH1 port with DHCP and ETH2 port with static IP 192.168.0.10.

All PRI Mediatrix gateways are pre-configured on ETH5 with DHCP and ETH1-4 port with static IP 192.168.0.10.

It is therefore important to have the DHCP server on the network. Both scenarios are described here:

With DHCP server: look for the IP address assigned to the gateway by using a sniffer or by consulting the DHCP server log. Using this address it is possible to enter the admin area and configure the static IP address for the gateway.

Without DHCP server: assign the emergency static IP address using a **partial reset** of the equipment using the following procedure:

- 1) Turn on the equipment and wait until it is working (the LED Power stops flashing)
- 2) With a fine/strong object press the RESET/ DEFAULT button. Wait until all the LED buttons flash (before only the LED POWER button flashes and after 5-7 seconds all the others). As soon as all the LED buttons flash, release the RESET/ DEFAULT button. After this partial reset the gateway will start with the static IP address: **192.168.0.1**. Make sure therefore that there is no other equipment with the same address on the same network, or connect the PC directly to the gateway (you don't require a crossed cable).

Note: Only press the reset/ default button for between 7 and 11 seconds. Release the button before the LED light stops flashing; otherwise the equipment will follow a complete reset.

Once you have the IP address of the gateway open a browser and contact the gateway. The username is **public** (no password).

Changing the DNS and Gateway (Menu Network – Host) settings



System
Network
ISDN
SIP
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Status
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Host

Automatic Configuration Interface	
Automatic IPv4 config source network:	<input type="text" value="Uplink"/>
Automatic IPv6 config source network:	<input type="text" value="UplinkV6"/>

Host Name Configuration	
Domain Name	
Configuration Source:	<input type="text" value="Automatic IPv4"/>
Domain Name:	<input type="text"/>
Host Name	
Host Name:	<input type="text"/>

Default Gateway Configuration	
IPv4	
Configuration Source:	<input type="text" value="Static"/>
Default Gateway:	<input type="text" value="10.0.1.138"/>
IPv6	
Configuration Source:	<input type="text" value="Automatic IPv6"/>
Default Gateway:	<input type="text"/>

DNS Configuration	
Configuration Source:	<input type="text" value="Automatic IPv4"/>
Primary DNS:	<input type="text"/>
Secondary DNS:	<input type="text"/>
Third DNS:	<input type="text"/>
Fourth DNS:	<input type="text"/>

SNTP Configuration	
Configuration Source:	<input type="text" value="Static"/>
SNTP Host:	<input type="text" value="europe.pool.ntp.org:123"/>
Synchronization Period:	<input type="text" value="1440"/>
Synchronization Period On Error:	<input type="text" value="60"/>

Time Configuration	
Static Time Zone:	<input type="text" value="EST5EDT4,M3.2.0/02:00:00,M11.1."/>

To set a static IP address, set the **Configuration source** field to **Static**.

Insert the IP static address of the gateway and of the DNS (up to 4) on the *Gateway and Primary Secondary DNS* fields.

Server SNTP (synchronise time): insert a NTP server (either: **Europe.pool.ntp.org: 123** or **time.nist.org:123**)

Time Zone: set the value **WEST-1DWEST-2,M3.5.0/02:00:00,M10.5.0/03:00:00**

Modifying the IP settings (Menu Network – Interfaces)

Mediatrix

System Network ISDN SIP Media Telephony Call Router Man

Status Host Interfaces VLAN QoS Local Firewall IP Routing Network Firewall NAT

Interfaces

Name	Link	Type	Static IP Address	Static Default Router	Activation	
Lan1	eth2	IPv4 DHCP	10.0.0.220/24	10.0.0.254	Enable	-
Uplink	eth1	IPv4 Static	10.0.0.220/24	10.0.0.254	Enable	-
UplinkV6	eth1	IPv6 Auto-Conf			Disable	-
						+

To set or modify the IP address of the equipment enter the menu Network – Interfaces.

On the **Uplink line**, select **eth1, IPv4 Static** and set the IP address in the **Static IP Address** field.

if there is a problem with the routing in Sylog you will see:

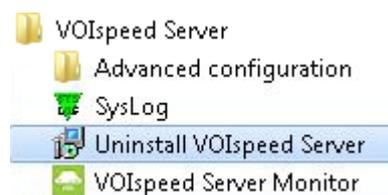
```
CallRouteTable [1F9B] Handle Call 34-35.
CallRouteTable [1F9C] Source interface isdn-Bri3 doesn't match source criteria sip-default
CallRouteTable [1F9D] Source interface isdn-Bri3 doesn't match source criteria isdn-Bri1
CallRouteTable [1F9E] Source interface isdn-Bri3 doesn't match source criteria isdn-Bri2
```

Syslog settings

Configure the Syslog Remote server (Menu System – Syslog)

For debugging purposes set the **Remote host (PBX Server)** address with the IP address of the syslog server, to be used in an emergency.

The syslog software is located under server installation folder.



Remember to see how the ISDN number is present by the provider, on the log file will be number@IPPBX on the Syslog you will see:

SipSignaling [07DB] > INVITE sip:624400@10.0.0.5:5058 SIP/2.0

if the number it is not the same on the PBX company number configuration the mediatrix will not be able to route the incomings calls.

Syslog Configuration	
Remote Host:	<input type="text" value="192.168.0.20"/>
Service Severity	
Authentication, Authorization and Accounting (AAA):	<input type="text" value="Warning"/>
Basic Network Interface (BNI):	<input type="text" value="Debug"/>
Call Routing (CROUT):	<input type="text" value="Warning"/>
Certificate Manager (CERT):	<input type="text" value="Warning"/>
Command Line Interface (CLI):	<input type="text" value="Warning"/>
Configuration Manager (CONF):	<input type="text" value="Warning"/>
Device Control Manager (DCM):	<input type="text" value="Warning"/>
Endpoint Administration (EPADM):	<input type="text" value="Warning"/>
Endpoint Services (EPSERV):	<input type="text" value="Warning"/>
Ethernet Manager (ETH):	<input type="text" value="Warning"/>
File Manager (FILE):	<input type="text" value="Warning"/>
Firmware Pack Updater (FPU):	<input type="text" value="Warning"/>
Host Configuration (HOC):	<input type="text" value="Warning"/>
IP Routing (IPROUTING):	<input type="text" value="Warning"/>
IP Synchronization (IPSYNC):	<input type="text" value="Warning"/>
Integrated Services Digital Network (ISDN):	<input type="text" value="Warning"/>
Local Quality Of Service (LQOS):	<input type="text" value="Warning"/>
Local Firewall (LFW):	<input type="text" value="Warning"/>
Media IP Transport (MIPT):	<input type="text" value="Warning"/>
Music On Hold (MOH):	<input type="text" value="Warning"/>
Notifications and Logging Manager (NLM):	<input type="text" value="Warning"/>
Process Control Manager (PCM):	<input type="text" value="Warning"/>
Service Controller Manager (SCM):	<input type="text" value="Warning"/>
SIP Endpoint (SIPEP):	<input type="text" value="Debug"/>
Simple Network Management Protocol (SNMP):	<input type="text" value="Warning"/>
Telephony Interface (TELIF):	<input type="text" value="Warning"/>
Web (WEB):	<input type="text" value="Warning"/>
Technical Assistance Centre	
Diagnostic Traces:	<input type="text" value="Enable"/>
Filter:	<input type="button" value="Edit"/>

Diagnostic Traces

Module	Traces					
+ Call Router	All	Info	Warning	Error	Critical	Disable
+ POTS	All	Info	Warning	Error	Critical	Disable
+ ISDN	All	Info	Warning	Error	Critical	Disable
+ CAS	All	Info	Warning	Error	Critical	Disable
+ Line	All	Info	Warning	Error	Critical	Disable
+ SIP	All	Info	Warning	Error	Critical	Disable
+ Stream	All	Info	Warning	Error	Critical	Disable
+ System	All	Info	Warning	Error	Critical	Disable

Configuring ISDN (Menu ISDN Basic Rate Interface)

Use the configuration default settings in the screen shot below:

Very Important Connection Type: ISDN sometime uses Point to Point, if not set connection fails.

Basic Rate Interface

Select Interface:

Interface Configuration	
Endpoint Type:	<input type="text" value="TE"/>
Clock Mode:	<input type="text" value="Slave"/>
Monitor Link State:	<input type="text" value="Disable"/>
Connection Type:	<input type="text" value="Point To Multipoint"/>
Signaling Protocol:	<input type="text" value="DSS1"/>
Network Location:	<input type="text" value="User"/>
Preferred Encoding Scheme:	<input type="text" value="G.711 a-Law"/>
Fallback Encoding Scheme:	<input type="text" value="G.711 u-Law"/>
Channel Allocation Strategy:	<input type="text" value="Ascending"/>
Maximum Active Calls:	<input type="text" value="2"/>
Signal Information Element:	<input type="text" value="Disable"/>
Inband Tone Generation:	<input type="text" value="Enable"/>
Inband DTMF Dialing:	<input type="text" value="Enable"/>
Overlap Dialing:	<input type="text" value="Enable"/>
Calling Name Max Length:	<input type="text" value="0"/>
Exclusive B-Channel Selection:	<input type="text" value="Disable"/>
Sending Complete:	<input type="text" value="Enable"/>
Send Restart On Startup:	<input type="text" value="Enable"/>
Link Establishment:	<input type="text" value="Permanent"/>
Hook-Flash Keypad:	<input type="text"/>
Accepted Status Causes:	<input type="text"/>
Accepted Progress Causes:	<input type="text" value="1-127"/>
Send Isdn Progress:	<input type="text" value="Send All"/>
Send Progress Indicator IE:	<input type="text" value="Send All"/>
TEI Negotiation:	<input type="text" value="Power Up"/>
Default TON for Calling Party Number IE:	<input type="text" value="Unknown"/>
Default NPI for Calling Party Number IE:	<input type="text" value="Unknown"/>
Default PI for Calling Party Number IE:	<input type="text" value="Presentation Allowed"/>
Default SI for Calling Party Number IE:	<input type="text" value="Context Dependent"/>
Default TON for Called Party Number IE:	<input type="text" value="Unknown"/>
Default NPI for Called Party Number IE:	<input type="text" value="Unknown"/>
Notification User Suspended:	<input type="text" value="Ignore"/>

Remember to apply changes in all connections:

Apply To The Following Interfaces				Check All	Uncheck All
<input checked="" type="checkbox"/> Bri1	<input type="checkbox"/> Bri2	<input type="checkbox"/> Bri3	<input type="checkbox"/> Bri4		

Submit

Primary Rate Interface

Select Interface:

Interface Configuration	
Line Type: [Configure]	E1
Endpoint Type:	<input type="text" value="TE"/>
Clock Mode:	<input type="text" value="Auto"/>
Port Pinout:	<input type="text" value="Auto"/>
Monitor Link State:	<input type="text" value="Enable"/>
Line Coding:	<input type="text" value="HDB3"/>
Line Framing:	<input type="text" value="CRC4"/>
Signaling Protocol:	<input type="text" value="DSS1"/>
Network Location:	<input type="text" value="User"/>
Preferred Encoding Scheme:	<input type="text" value="G.711 a-Law"/>
Fallback Encoding Scheme:	<input type="text" value="G.711 u-Law"/>
Channel Range:	<input type="text" value="1-30"/>
Channel Allocation Strategy:	<input type="text" value="Ascending"/>
Maximum Active Calls:	<input type="text" value="30"/>
Signal Information Element:	<input type="text" value="Disable"/>
Inband Tone Generation:	<input type="text" value="Enable"/>
Inband DTMF Dialing:	<input type="text" value="Enable"/>
Overlap Dialing:	<input type="text" value="Enable"/>
Calling Name Max Length:	<input type="text" value="0"/>
Exclusive B-Channel Selection:	<input type="text" value="Disable"/>
Sending Complete:	<input type="text" value="Enable"/>
Send Restart On Startup:	<input type="text" value="Enable"/>
Link Establishment:	<input type="text" value="Permanent"/>
Accepted Status Causes:	<input type="text"/>
Accepted Progress Causes:	<input type="text" value="1-127"/>
Send Isdn Progress:	<input type="text" value="Send All"/>
Send Progress Indicator IE:	<input type="text" value="Send All"/>

Configuring SIP (Menu SIP – Servers)

In this section the IP address of the VOIspeed 6 server is configured so that the Mediatrix gateway routes the incoming calls.

The fields to complete are:

Registrar Host: <IP_PBX_VOIspeed>:SIP Port

Proxy Host: <IP_PBX_VOIspeed>:SIP Port

Messaging server Host: <IP_PBX_VOIspeed>:SIP Port

Outbound Proxy Host: <IP_PBX_VOIspeed>:SIP Port

Note: Insert the IP address of the VOIspeed server following the portal used by the SIP interface (separated by a colon(:))(as in the example below). The default SIP portal uses the VOIspeed server and **5060 in any case need to match the SIP port that VOIspeed Server uses.**



The screenshot shows the Mediatrix configuration interface. The top navigation bar includes System, Network, ISDN, SIP, Media, Telephony, Call Router, and Managem. Below this, there are tabs for Gateways, Servers, Registrations, Endpoints, Authentication, Transport, Interop, and Misc. The 'Servers' tab is selected, and the 'SIP Default Servers' table is visible. The table has four rows, each with a label and a text input field containing the value '10.0.1.5:5060'.

SIP Default Servers	
Registrar Host:	10.0.1.5:5060
Proxy Host:	10.0.1.5:5060
Messaging Server Host:	10.0.1.5:5060
Outbound Proxy Host:	10.0.1.5:5060

Define the settings for the missing Zero (Menu Call Router – Route Config)

When the telephone operator identifies the type of call by inserting the TON (Type of Number) flag, the Mediatrix **suppresses** the initial zero from the national call numbers. The result is that all the numbers of the companies called show VOIspeed without the initial zero. Since the contact numbers in the phone book begin with zero, this prevents seeing their number. Normally this only happens with **Telecom**, whilst with other operators (for example **Fastweb**)(English companies?) the problem doesn't exist, in which case please proceed to the next chapter.

Solutions to the missing zero on numbers called on the incoming number:

1. Add a new type of mapping on **Mapping type** (click on the + button) and select the fields set out in the screen shot below. Then click on **Submit and Insert Expression**.

Mapping Type

Configure Mapping Type 1	
	Value
Name	<input type="text" value="AddZero"/>
Criteria	<input type="text" value="Calling E164"/>
Transformation	<input type="text" value="Calling E164"/>
Config Status	

Submit Submit And Insert Expression Cancel

- A window will pop up with a new mapping type to be inserted (Mapping Express section). Select the fields and insert the data shown in the screen shot below: Then click on **Submit and Insert Expression**.

BRI

Mapping Expression

Configure Mapping Expression 1		
	Value	Suggestion
Type	Calling E164 to Calling E164	
Name	<input type="text" value="AddZero"/>	<input type="text" value="--- Suggestion ---"/>
Criteria	<input type="text" value=".,+"/>	<input type="text" value="--- Suggestion ---"/>
Transformation	<input type="text" value="0\0"/>	<input type="text" value="--- Suggestion ---"/>
Sub Mappings	<input type="text"/>	<input type="text" value="--- Suggestion ---"/>
Config Status		

PRI

Mapping Expression

Configure Mapping Expression 1		
	Value	Suggestion
Type	Calling E164 to Calling E164	
Name	<input type="text" value="AddZero"/>	<input type="text" value="--- Suggestion ---"/>
Criteria	<input type="text" value=".,+"/>	<input type="text" value="--- Suggestion ---"/>
Transformation	<input type="text" value="0&#92;0"/>	<input type="text" value="--- Suggestion ---"/>
Sub Mappings	<input type="text"/>	<input type="text" value="--- Suggestion ---"/>
Config Status		

Submit Submit And Insert Expression Cancel

- Add a new map on **Mapping type** (click on the button as in point 1). Select the fields and insert the data shown in the screen shot below: Then click on **Submit and Insert Expression**.

Mediatrix®

System Network ISDN SIP Media Telephony **Call Router** Management

Status Route Config Auto-routing

➤ Mapping Type

Configure Mapping Type 2	
	Value
Name	National
Criteria	Calling TON
Transformation	None
Config Status	

Submit Submit And Insert Expression Cancel

4. A window will pop up to create a new expression on the mapping type as soon as this is inserted Mapping Expression section). Select the fields and insert the data shown in the screen shot below: Then click on **Submit and Insert Expression**.

Mediatrix®

System Network ISDN SIP Media Telephony **Call Router** Management

Status Route Config Auto-routing

➤ Mapping Expression

Configure Mapping Expression 2		
	Value	Suggestion
Type	Calling TON to None	
Name	National	--- Suggestion ---
Criteria	national	--- Suggestion ---
Transformation		--- Suggestion ---
Sub Mappings	AddZero	--- Suggestion ---
Config Status		

Submit Submit And Insert Expression Cancel

Until now we have concentrated on the settings for the Mediatrix on incoming national calls. We now need to instruct the Mediatrix to follow these settings with the routing rules.

Missing Zero on company numbers called. The incoming calls arrive without the zero also for company numbers. If you want to change this rule, you just need to follow the same steps but on the rule for routing substitute the criteria “ Calling E.164” with **Called E.164** and “Calling TON” with **Called TON**. Obviously you need also to use a differed name for the rule.

Anonymous calling (Menu Call Router – Route Config)

To set up anonymous calling for outgoing calls, insert in the Sip ID of the gateway configuration of the gateway in V6, the code that leaves the operator to go out anonymously.

For Telecom this code is either `*#373#` and code `*67#`

Anonymous calling (Menu Call Router – Route Config)

This setting changes ALL the calls leaving the gateway. **If anonymous calling is required from a particular base, create a new Gateway and link access to the base required.**

Show a chosen number (Menu Call Router – Route Config)

To show a chosen number from a multi number block, there are no setting changes on Mediatix. Instead set the SipIP of the ISDN gateway in VOISpeed 6 to the chosen exit number (this number will appear in the field from the sender and will be used by Mediatix as exit number). It is necessary however, that the number is configured on the ISDN line otherwise this setting will be ignored and the call will leave with the number at the top of the list.

Defining Routing Rules (Menu Call router – Rouge Config)

The routing rules define the route used by the gateway to direct the incoming and outgoing calls: their creation and definition are therefore determined by the correct working of the equipment. The routing rules connect the telephone interface (PRI, ISDN or FSX) to the SIP interface (gateway) by routing the calls coming from the telephone network towards the VOISpeed server, and by directing to the telephone lines the calls coming from the PBX.

On the Route Config menu there is a list of the routes configured. The top of the list is obviously empty.



Mediatix®

System Network ISDN SIP Media Telephony Call Router M

Status Route Config Auto-routing

Route Config

Config Modified: no

Route	Index	Sources	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions
	1	sip-Gateway2	None				hunt-ISDN	Edit [v] + -
	2	isdn-Bri1, isdn-Bri2, isdn-Bri3, isdn-Bri4	None		National, NationalCalled	EarlyDisconnect	sip-Gateway2	Edit [^] + -

Very Important Check Source and destination need to match the sip gateway you created before, in this case (sip-Gateway2) edit route and use the suggestion option

Route

Configure Route 1		Suggestion
	Value	
Sources	sip-Gateway2	--- Suggestion --- --- Suggestion --- sip-Gateway2 sip-default isdn-Bri1 isdn-Bri2 isdn-Bri3 isdn-Bri4 route-
Properties Criteria	None	
Expression Criteria		
Mappings		
Signaling Properties		
Destination	hunt-ISDN	
Config Status		

Submit Cancel

1. Configuring a single BRI

Click on the button + to reach the new rule for incoming calls. Select the settings in the screen shot below: Then click on Submit.

Route

Configure Route 1		Suggestion
	Value	
Sources	isdn-Slot2/E1T1	--- Suggestion ---
Properties Criteria	None	
Expression Criteria		--- Suggestion ---
Mappings	National	--- Suggestion ---
Signaling Properties	EarlyDisconnection	--- Suggestion ---
Destination	sip-default	--- Suggestion ---
Config Status		

Submit Cancel

Use this method for all the other ISDN interfaces. (isdn – Bri2,3,4) or PRI (slot2/E1T1).

Then configure the rule for routing the exit calls. The only parameters to configure are as follows:

Sources: select sip-default

Destination: select isdn-Bri1

As shown in the screen shot below.

Click on **Submit**

Route

Configure Route 2		Value	Suggestion
Sources	<input type="text" value="sip-default"/>		--- Suggestion ---
Properties Criteria	<input type="text" value="None"/>		
Expression Criteria	<input type="text"/>		--- Suggestion ---
Mappings	<input type="text"/>		--- Suggestion ---
Signaling Properties	<input type="text"/>		--- Suggestion ---
Destination	<input type="text" value="isdn-Slot2/E1T1"/>		--- Suggestion ---
Config Status			

Submit Cancel

The administration of the PBX can decide how to route the calls leaving based on two criteria:

- 1) Route the calls directly and independently to each base access. This mode is useful in cases in which you want to create different routing rules to specifically route ISDN lines (and especially in the case of a multi office phone system where the ISDN resources are separate and not shared). Every base access (BRI) will therefore be directed via another SIP gateway which should be configured on the LCR of the VOIspeed server.
- 2) Route the calls using the ISDN resources as an integral line. In this case the BRI interfaces will be grouped in a hunt group and used as a single resource on the gateway which will be utilised according to how busy they are. The BRI interface hunt group will be shown as a single gateway on the VOIspeed PBX that will be not be visible on the single BRI.
- 3) When you have more base rate interfaces (BRIs) (i.e. 4 or more) you can apply both methods above (1 and 2).

2. Configuring a second separate BRI

This setting enables the routing of direct phone calls independently on every base access. To this you need to create a SIP gateway for every access base.

Creating a new gateway

To create a new gateway and assign to a BRI interface enter the **SIP –Gateways** menu. On the *Gateway Configuration* write in the field Name the name of the new gateway (for example Gateway 2) and click on the + button.

Some changes require to restart a service to apply new configuration.
Please click this link to access the services table.

Gateways

Gateway Status						
Name	Signaling Network	Media Networks	Port	Secure Port	State	
default	Uplink	Uplink	0	0	Ready	

Gateway Configuration						
Name	Signaling Network	Media Networks	Media Networks Suggestion	Port	Secure Port	
default	Uplink	Uplink	--- Suggestion ---	0	0	-
gateway2						+

Once the gateway appears, chose *Uplink* in the **Media Networks Suggestion** and assign a **different** SIP port to the one used by the sip-default gateway (for example, using 5062).

Then create a routing rule which links the SIP gateway to the ISDN interface, following the instructions under **Configuring a single BRI** (paragraph above), making sure you select the BRI interface.

Sources: insert *gateway 2*

Destination: insert *isdn-Bri2*

For additional BRI interfaces apply the same criteria, to use a different SIP port for each new gateway created. This port should be set in the Proxy field of the SIP gateway setting on the VOISpeed 6 PBX.

3. Configuring a second BRI when shared in a HUNT group.

In this case all or part of the BRI interfaces will be grouped in a hunt group and used as a single gateway resource that will be utilised according to their occupancy. The hunt group of the BRI interfaces will be shown as a single gateway on the VOISpeed PBX.

Creating a hunt group

From the menu **Call router – Route Config** create a hunt group by clicking on the + sign.

Assign the group name and select from the drop down menu *Suggestion* the BRI interface to be included in the group. Leave all the other settings as default settings.

➤ **Hunt**

Configure Hunt End		
	Value	Suggestion
Name	<input type="text" value="Gruppo-isdn1"/>	
Destinations	<input type="text" value="isdn-Bri1, isdn-Bri2, isdn-Bri3, isdn-Bri4"/>	--- Suggestion ---
Selection Algorithm	<input type="text" value="Sequential"/>	
Timeout (seconds)	<input type="text" value="0"/>	
Causes	<input type="text" value="31, 34, 38, 41, 42, 43, 44, 47"/>	--- Suggestion ---
Config Status		

After having created the group, create an exit call routing rule from the sip-default gateway to the hunt group just created.

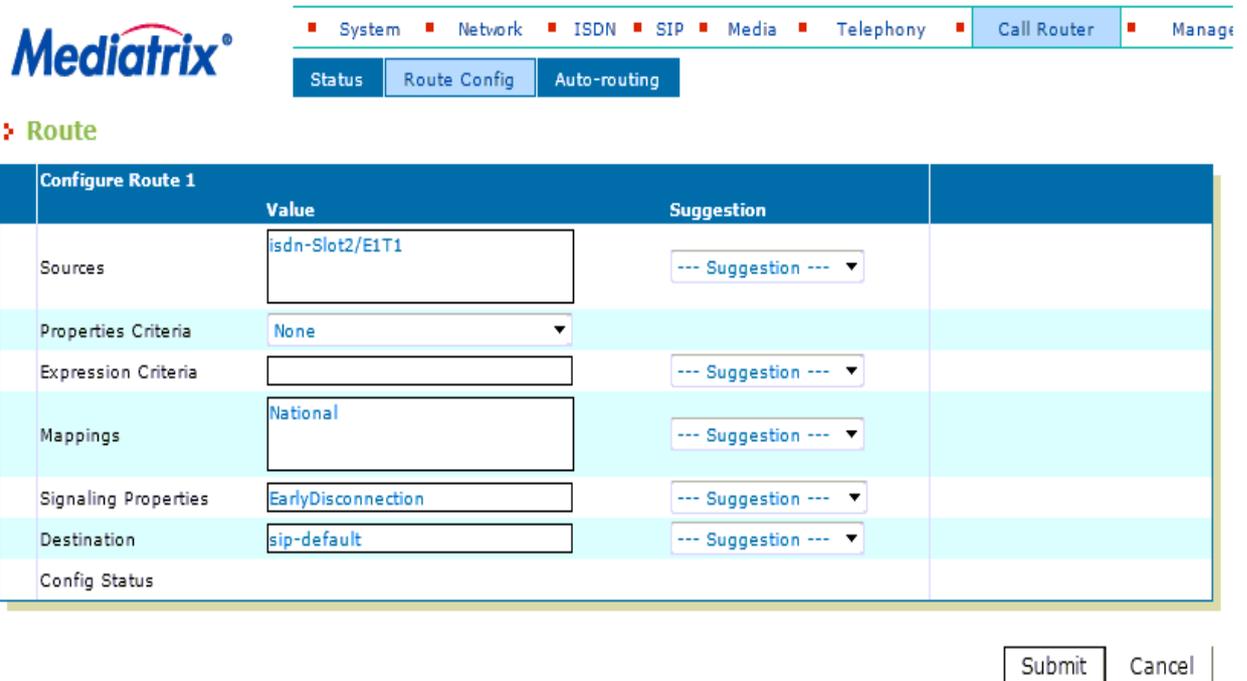
ISDN 30 seconds Timeout (Media Menu)

When the call is finished on ISDN, you can hear the disconnection tone for 30 seconds, after which the call will be disconnected. To avoid waiting for these 30 seconds, configure the Mediatrix as follows:

Configure Signaling Properties 1		
	Value	Suggestion
Name	<input type="text" value="EarlyDisconnection"/>	
Early Connect	<input type="text" value="Disable"/>	
Early Disconnect	<input type="text" value="Enable"/>	
Destination Host	<input type="text"/>	--- Suggestion ---
Allow 180 with SDP	<input type="text" value="Enable"/>	
Allow 183 without SDP	<input type="text" value="Enable"/>	
Privacy	<input type="text" value="Disable"/>	
SIP Headers Translations	<input type="text"/>	--- Suggestion ---
Call Properties Translations	<input type="text"/>	--- Suggestion ---
Config Status		

Press Submit to save the changes.

Then, in the routing rule from ISDN to SIP (i.e. for incoming calls), activate the rule just created in the 'signalling properties' field:



The screenshot shows the Mediatrix interface with the 'Call Router' menu item selected. The 'Route Config' tab is active, displaying the 'Configure Route 1' form. The form contains the following fields:

	Value	Suggestion
Sources	isdn-Slot2/E1T1	--- Suggestion ---
Properties Criteria	None	
Expression Criteria		--- Suggestion ---
Mappings	National	--- Suggestion ---
Signaling Properties	EarlyDisconnection	--- Suggestion ---
Destination	sip-default	--- Suggestion ---
Config Status		

At the bottom right of the form, there are 'Submit' and 'Cancel' buttons.

Media Settings (Media Menu)

1. Configuring Codec (Media Menu – Codecs)

Enable Voice and Data for G711 a-Law codec. All the other codecs will be disabled in both the voice and data sections.

Codecs

Select Endpoint: Default

Codec	Voice	Data	Advanced
G.711 a-Law	Enable	Enable	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	Disable		Edit
T.38		Disable	Edit
Clear Mode	Disable	Disable	Edit
Clear Channel	Disable	Disable	Edit
X CCD	Disable	Disable	Edit

In the Generic VAD section, you have to disable the VAD selecting Disable from the drop down menu showing the following setting:

Generic Voice Activity Detection (VAD)	
Enable (G.711 and G.726):	Disable

Click on Edit in the G711.a-Law window to open the advanced codec settings to set the packet length voice to 20ms as shown below:

Codecs

Select Endpoint: Default

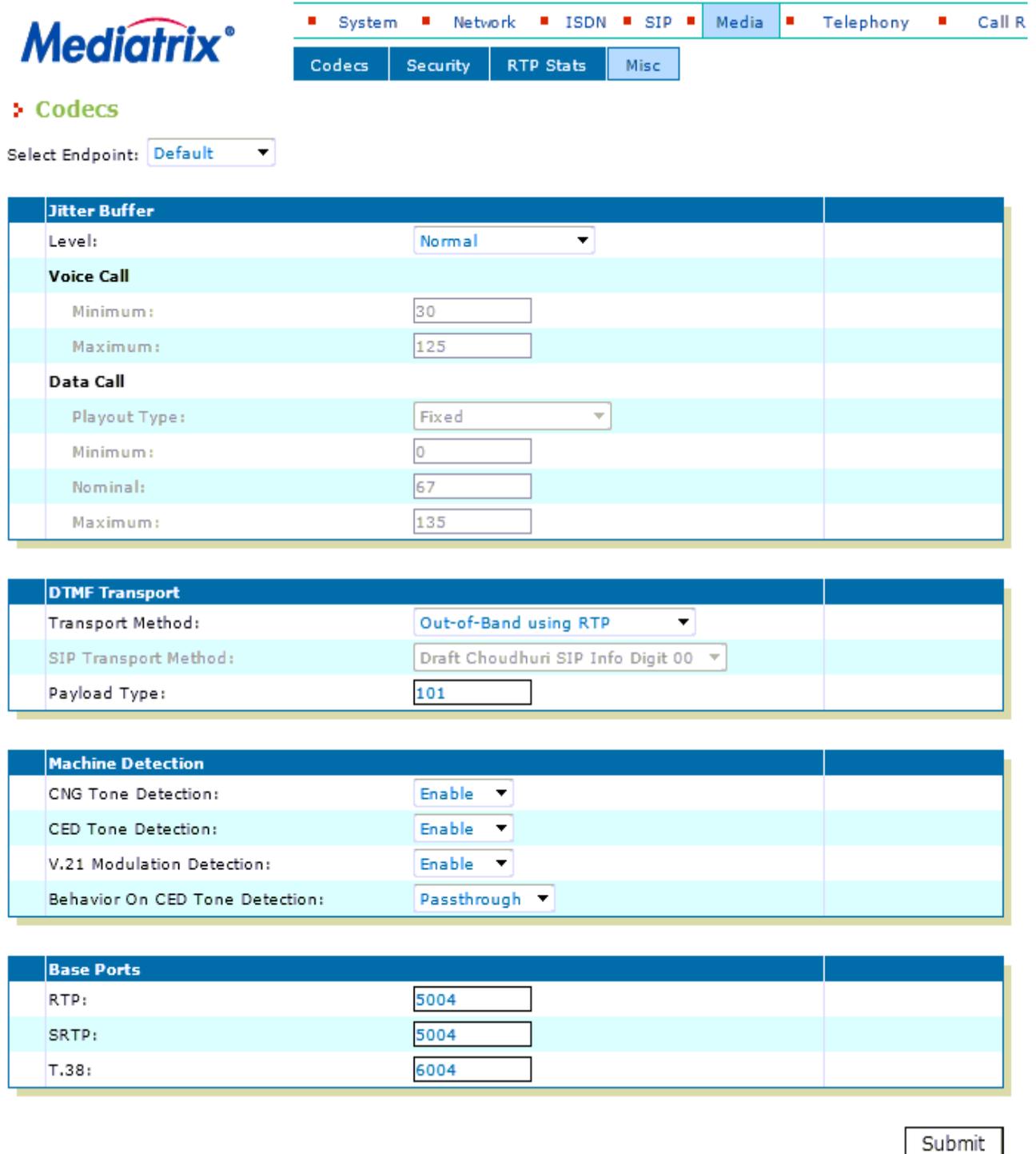
Select Codec: G.711 a-Law

G.711 a-Law	
Voice Transmission:	Enable
Voice Priority:	0
Data Transmission:	Enable
Data Priority:	0
Minimum Packetization Time:	20 ms
Maximum Packetization Time:	20 ms

Submit Cancel

2. Configuring DTMF (Media Menu – Misc)

Finally, configure also the part relating to the sending of the DTMF tone by inserting “Out-of-brand using RTP” in the section “Transport Method” whilst compiling the filed Payload Type with “101” value as shown in the screen shot below:



The screenshot shows the Mediatix configuration interface. The top navigation bar includes tabs for System, Network, ISDN, SIP, Media, Telephony, and Call R. The 'Media' tab is selected, and the 'Misc' sub-tab is active. The 'Codecs' section is expanded, showing a 'Select Endpoint' dropdown set to 'Default'. Below this are four configuration panels:

- Jitter Buffer:** Level is set to 'Normal'.
- Voice Call:** Minimum is 30, Maximum is 125.
- Data Call:** Playout Type is 'Fixed', Minimum is 0, Nominal is 67, Maximum is 135.
- DTMF Transport:** Transport Method is 'Out-of-Band using RTP', SIP Transport Method is 'Draft Choudhuri SIP Info Digit 00', and Payload Type is '101'.
- Machine Detection:** CNG Tone Detection, CED Tone Detection, and V.21 Modulation Detection are all 'Enable'. Behavior On CED Tone Detection is 'Passthrough'.
- Base Ports:** RTP is 5004, SRTP is 5004, and T.38 is 6004.

A 'Submit' button is located at the bottom right of the configuration area.

Ring tone settings (Telephony – Misc)

External phone calls might ring with a strange ringtone, not used in the UK. To remedy this, configure the Mediatrix as follows.

Enter in the Telephony –Misc section and select UK from the Country selection menu:

Then press Submit to apply the changes.



The screenshot shows the Mediatrix web interface. At the top left is the Mediatrix logo. A navigation bar contains several menu items: System, Network, ISDN, SIP, Media, Telephony, and a partially visible 'C'. Below this, a sub-menu is open for 'Telephony', showing options: DTMF Maps, Services, Tone Customization, Music on Hold, and Misc. The 'Misc' option is selected. Below the sub-menu, the heading 'Misc' is displayed. A table with a blue header row is shown. The header row is labeled 'Country'. The table contains one row with the label 'Country Selection:' and a dropdown menu currently displaying 'UK1'.

Country	
Country Selection:	UK1

Reboot the system (Menu Reboot)

Although it is possible restart services individually after each setting configured, it is advisable to reboot the gateway completely at the end of the entire configuration.