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.

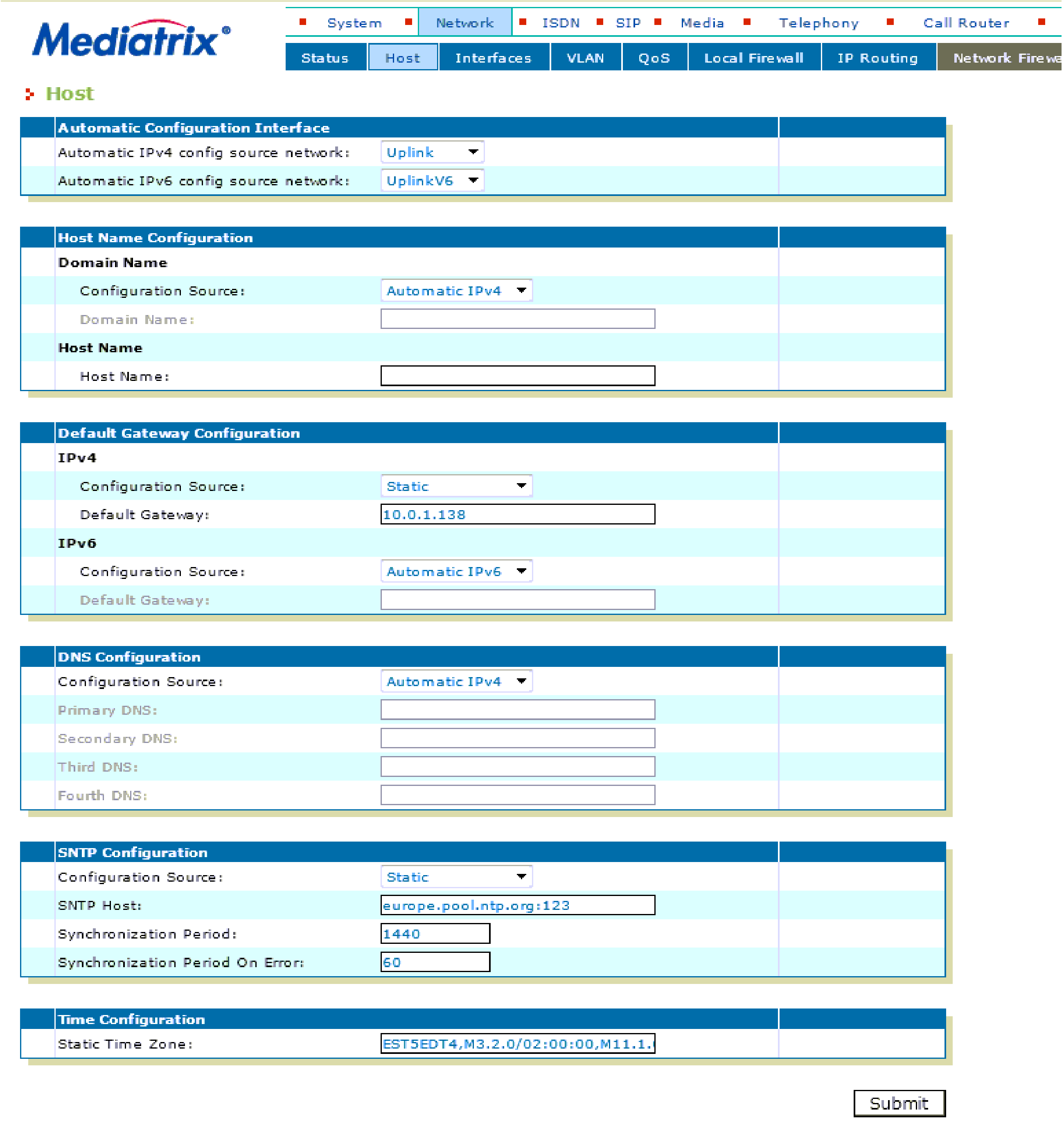
Without DHCP server: assign the emergency static IP address using a **partial reset** of the equipment using the following procedure:

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 with 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



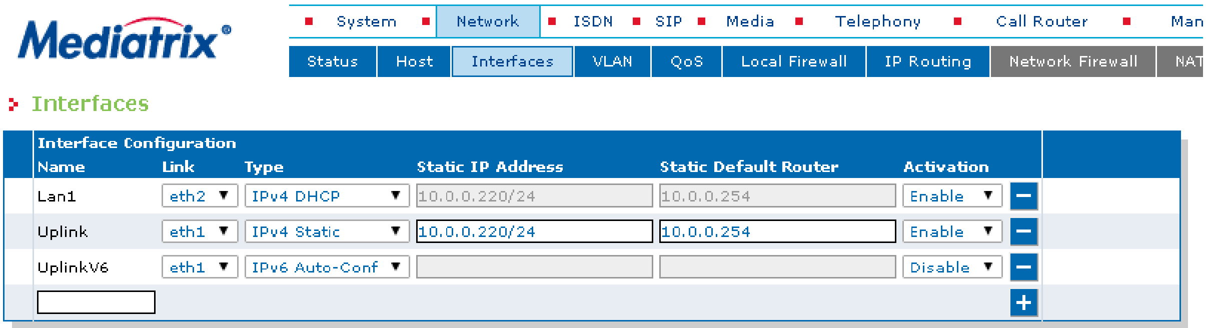
To set a static IP address, set the **Configuration source** field to **Static.**

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)



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

On the <u>**Uplink line**</u>, 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:

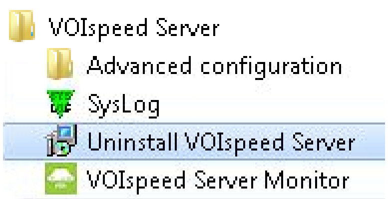
```
CallRouteTable [1F9B] Handle Call 34-35.
CallRouteTable [1F9C] Source interface isdn-Bri3 doesn't match source criteria sip-default
CallRouteTable [1F9D] Source interface isdn-Bri3 doesn't match source criteria isdn-Bri1
CallRouteTable [1F9E] Source interface isdn-Bri3 doesn't match source criteria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

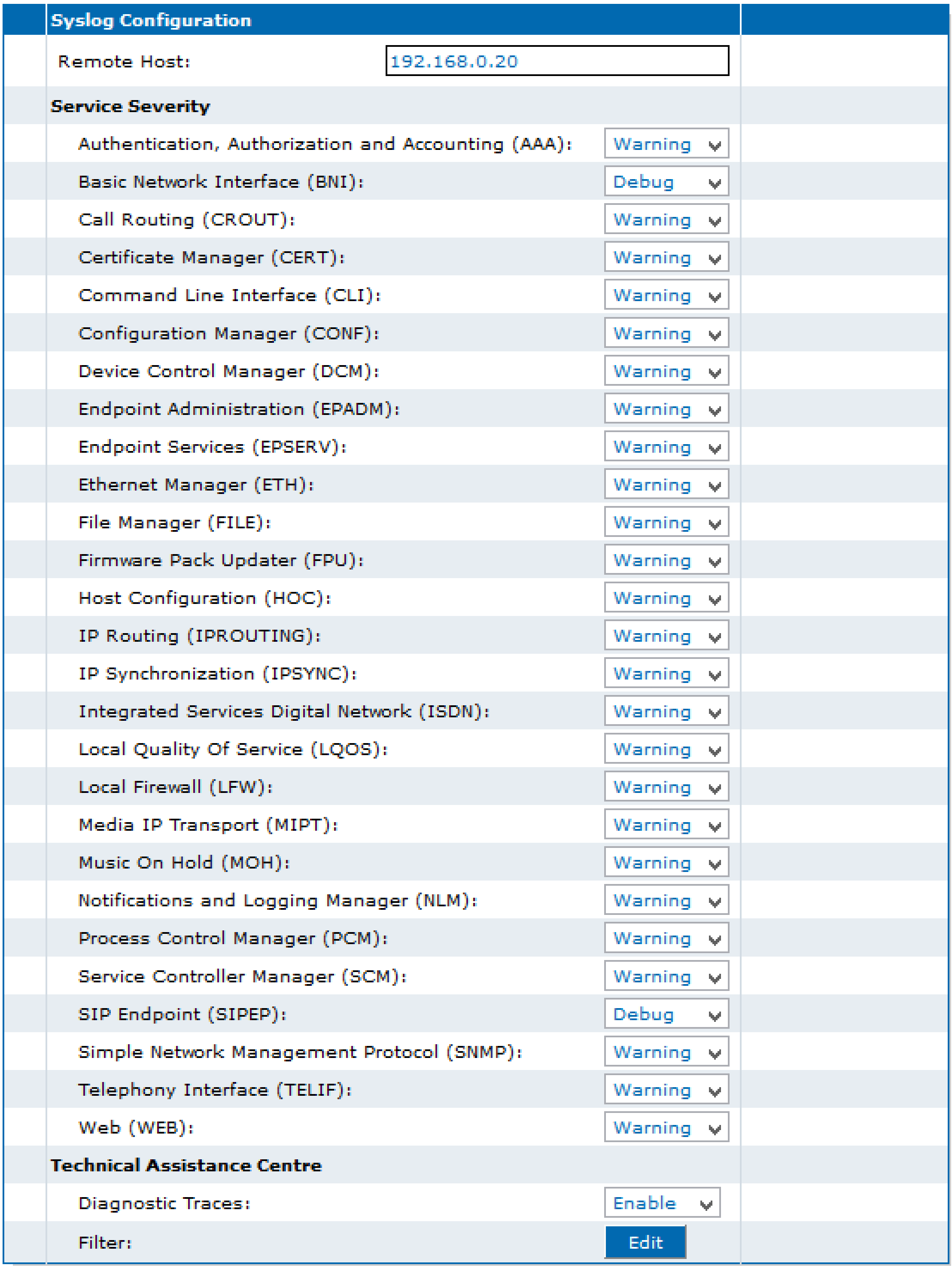
The syslog software is located under server installation folder.



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.



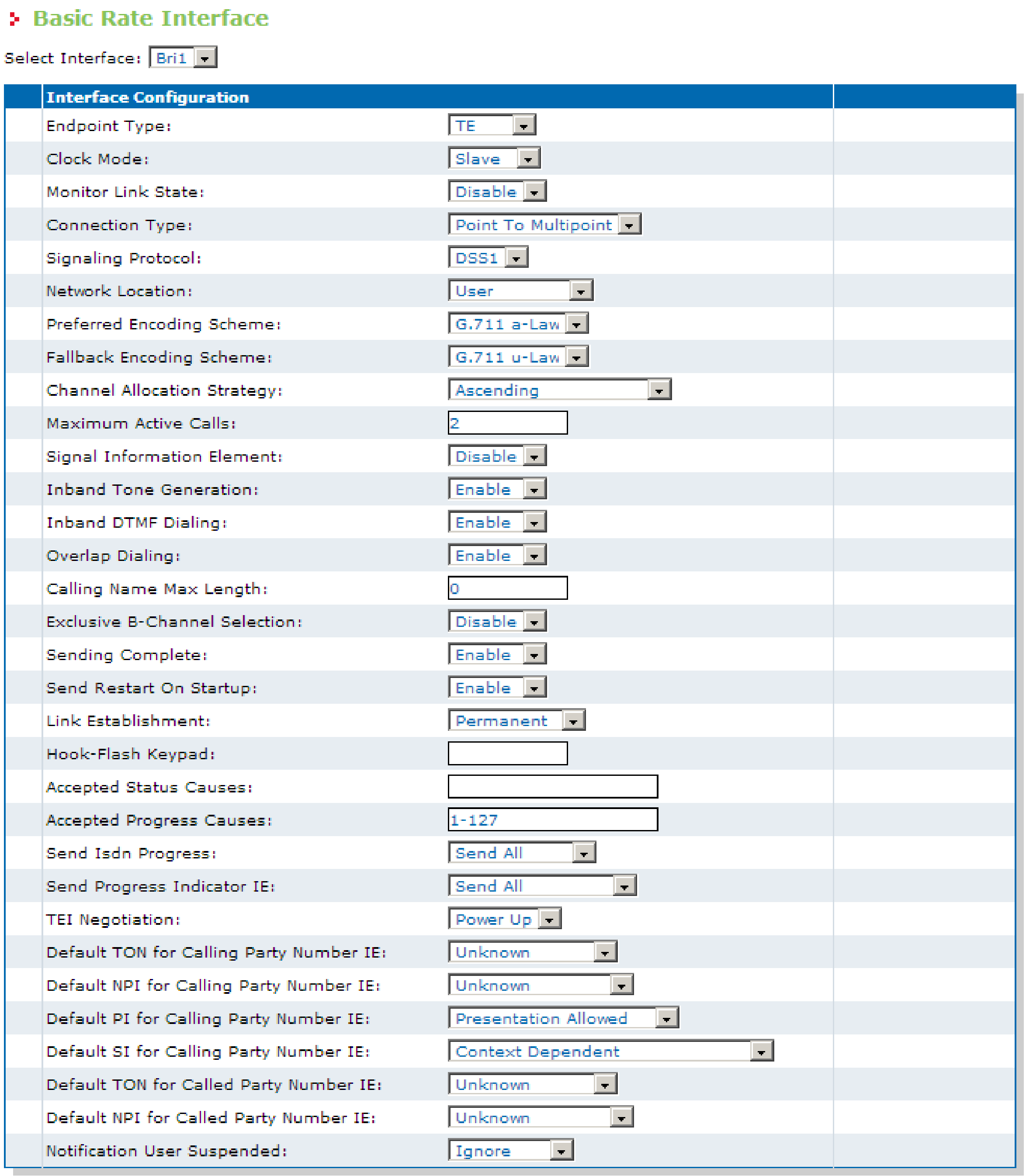
Diagnostic Traces

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: IDSN sometime uses Point to Point, if not set connection fails.

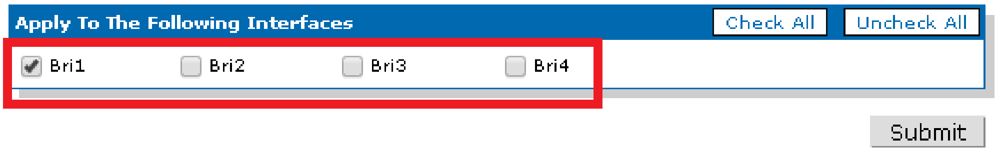


Basic Rate Interface

Select Interface: Bri1

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

Remember to apply changes in all connections:



VOI SPEED

Mediatrix®

System | Network | ISDN | SIP | Media | Telephony

Status | Primary Rate Interface | Interop | Timer | Services

Primary Rate Interface

Select Interface: Slot2/E1T1

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

Submit

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:

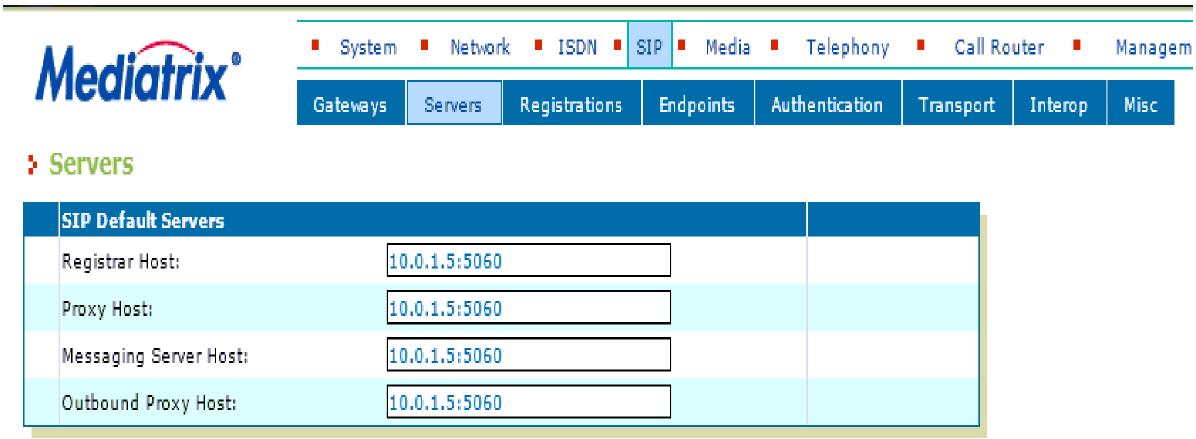
Registrar Host: <IP_PBX_VOIspeed>:SIP Port

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.**

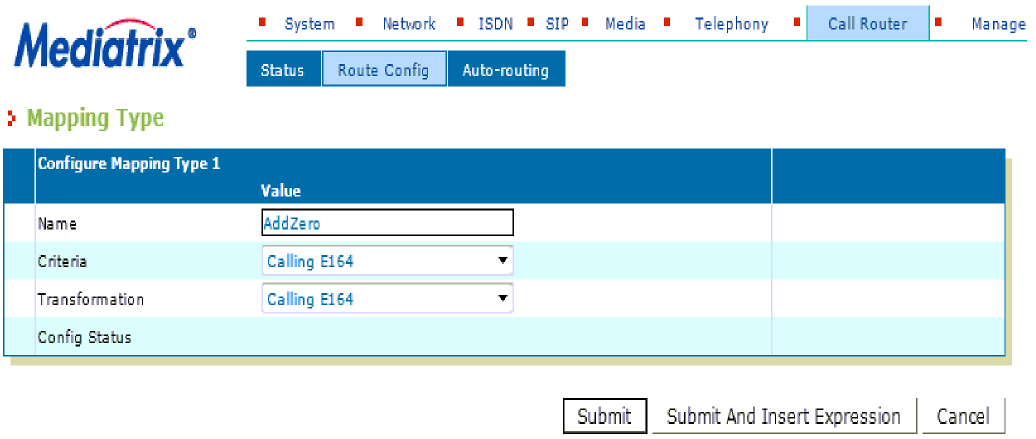


Define the settings for the missing Zero (Menu Call Router – Route Config)

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.

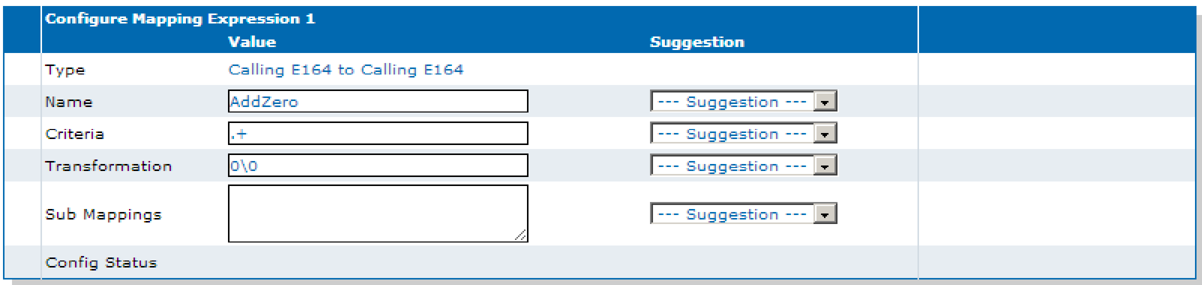
Solutions to the missing zero on numbers called on the incoming number:

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**.

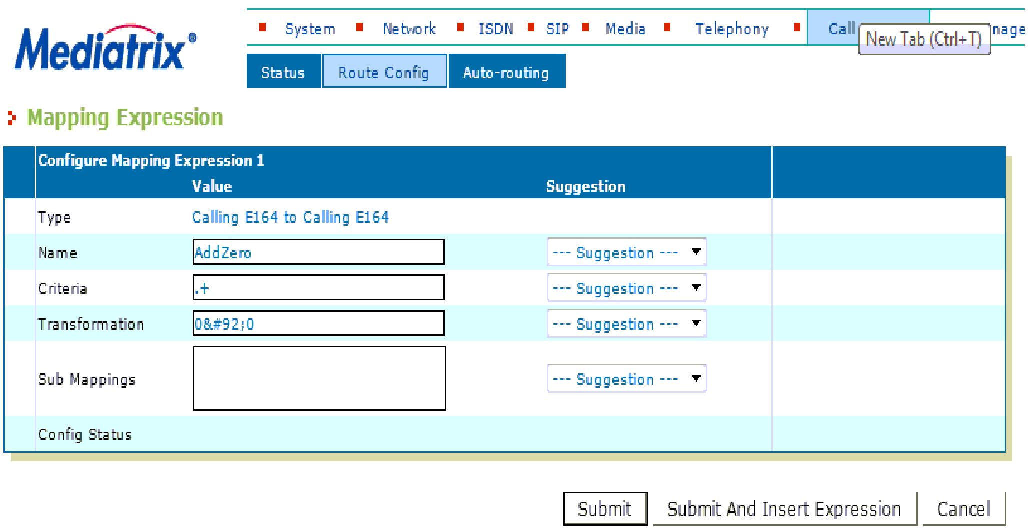


2. A window will pop up with a new mapping type to be inserted (Mapping Express section). Select the fields and insert the data shown in the screen shot below: Then click on **Submit and Insert Expression**.

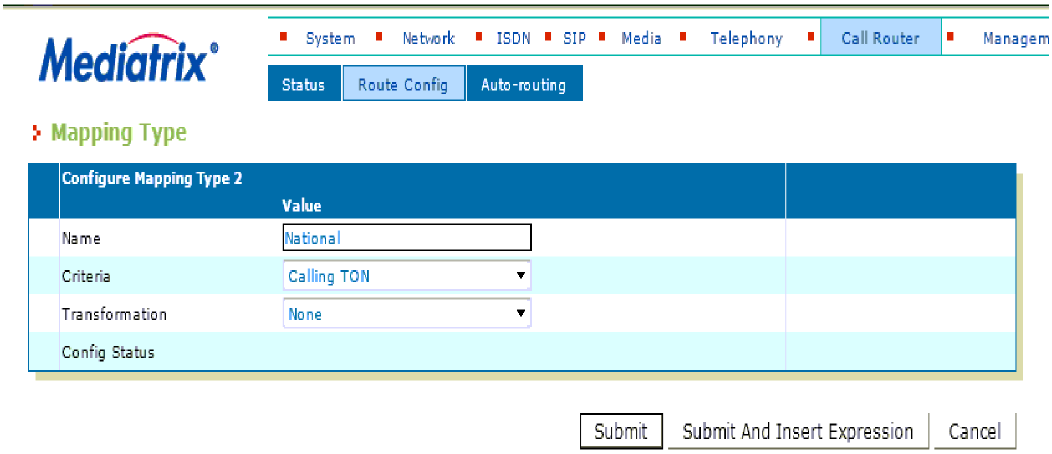
BRI



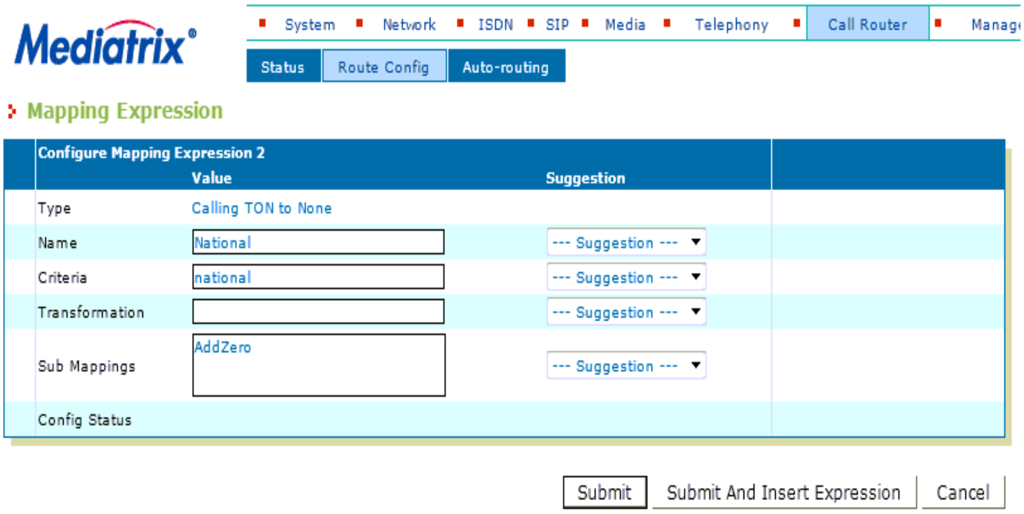
PRI



3. 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**.



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**.



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.

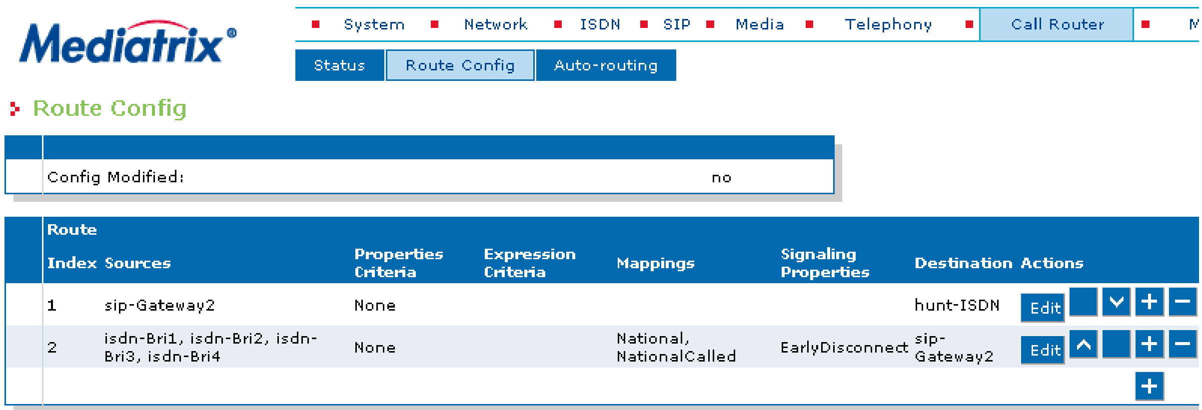
Show a chosen number (Menu Call Router – Route Config)

To show a chosen number from a multi number block, there are no setting changes on Mediatrix. 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 Mediatrix 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.

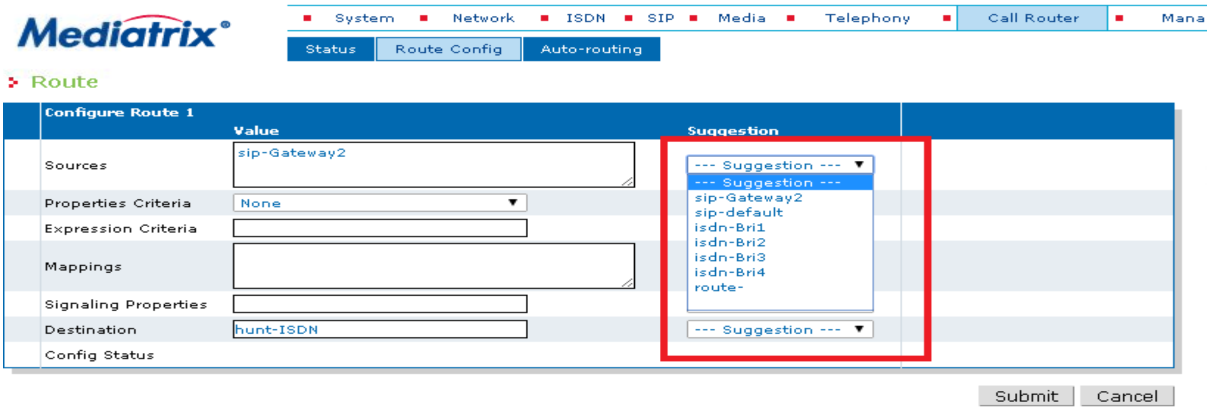
Defining Routing Rules (Menu Call router – Rouge Config)

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.

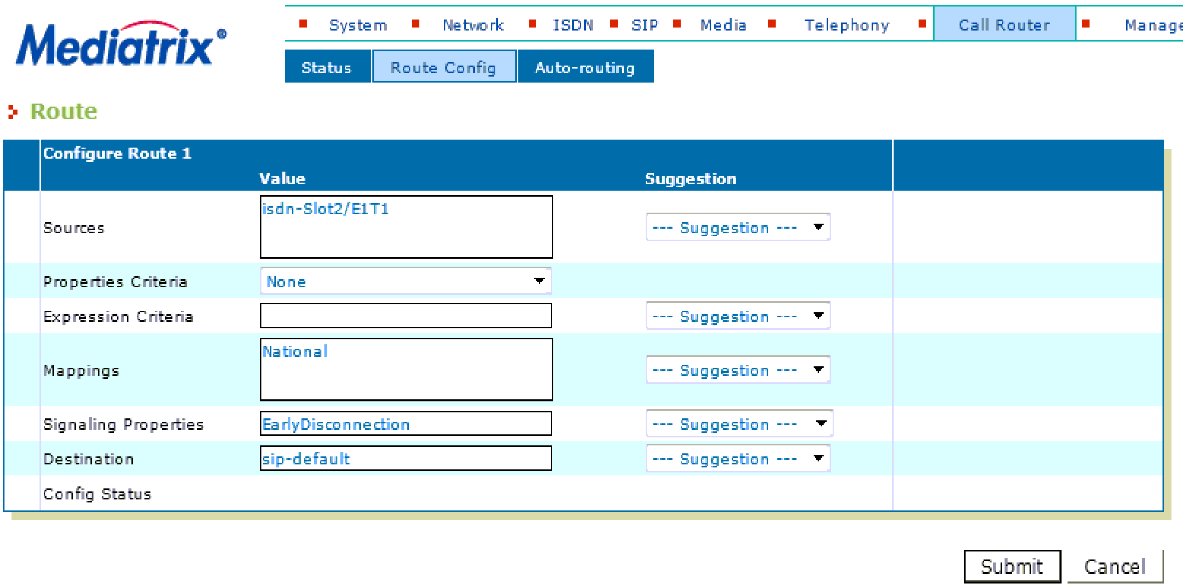


Very Important Check Source and destination need to match the sip gateway you created before, in this case (sip-Gateway2) edit route and use the suggestion option

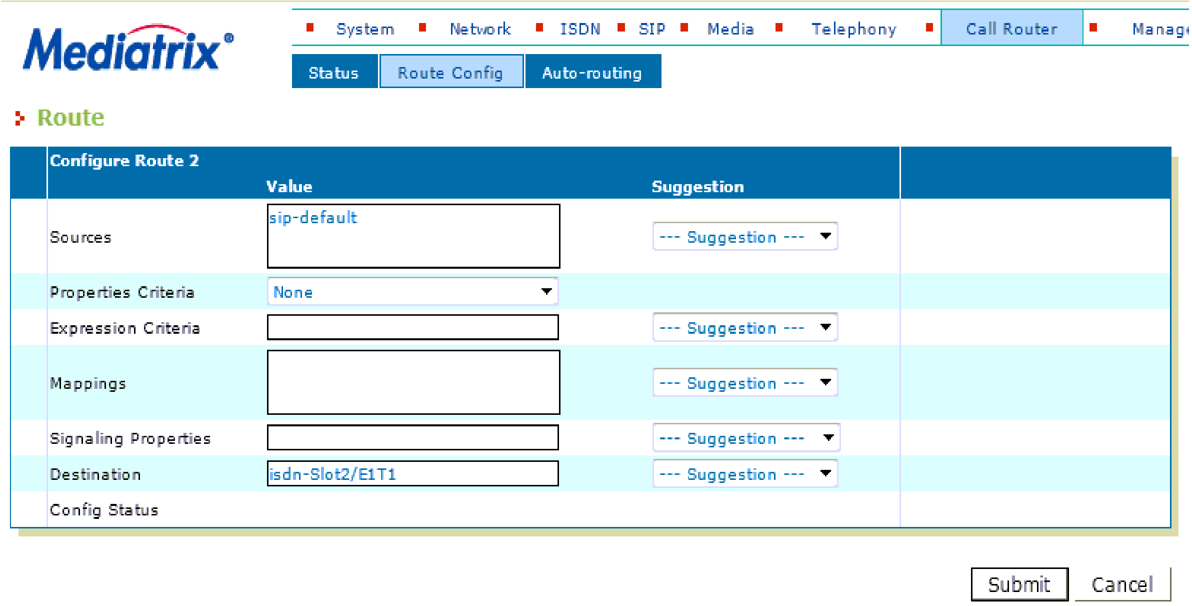


1. Configuring a single BRI

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



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**



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

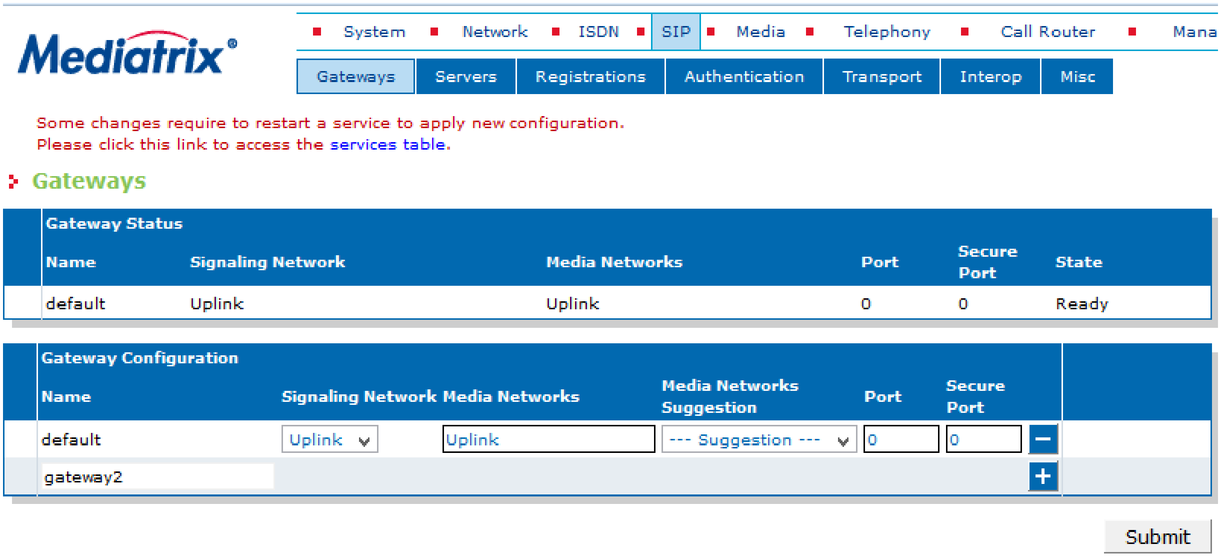
1) Route the calls directly and independently to each base access. This mode is useful in cases in which you want to create different routing rules to specifically route ISDN lines (and especially in the case of a multi office phone system where the ISDN resources are separate and not shared). Every 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.



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.

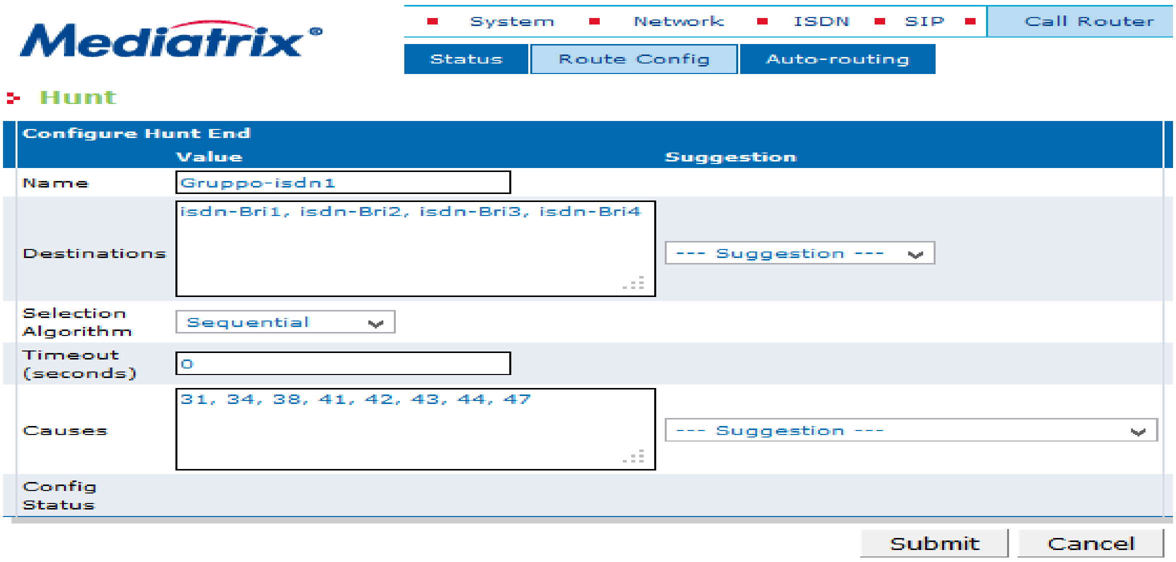
3. Configuring a second BRI when shared in a HUNT group.

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.

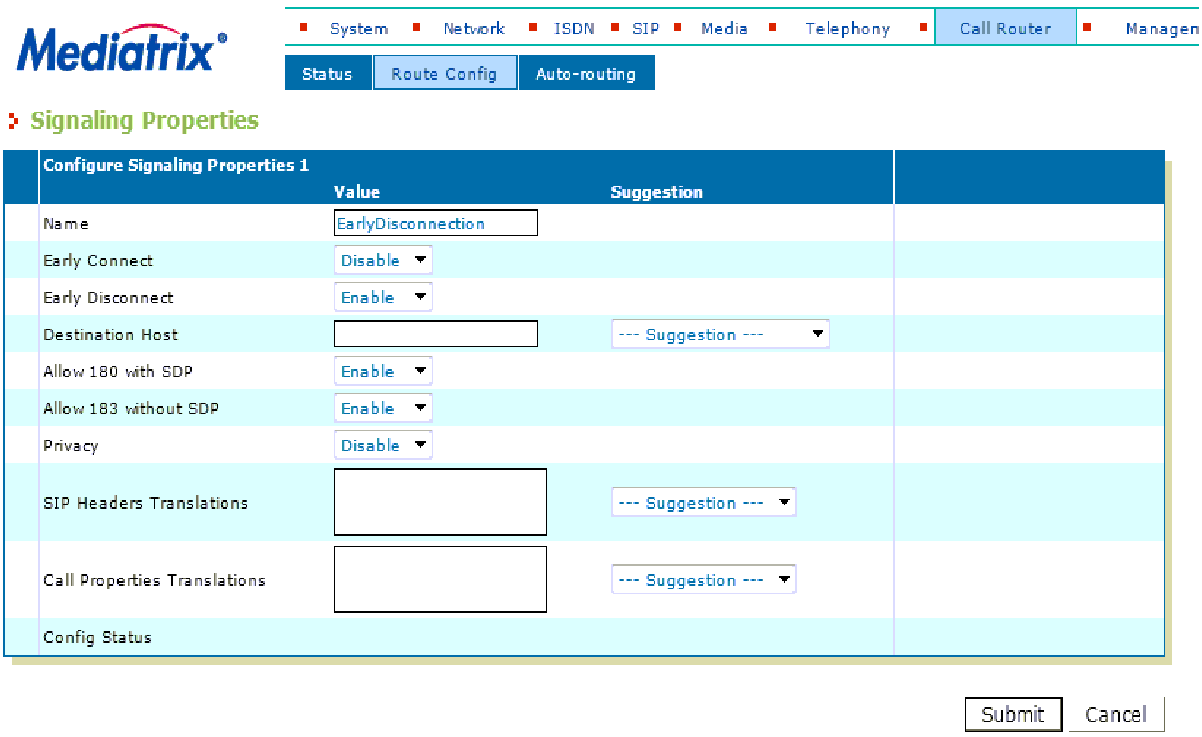
Assign the group name and select from the drop down menu *Suggestion* the BRI interface to be included in the group. Leave all the other settings as default settings.



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

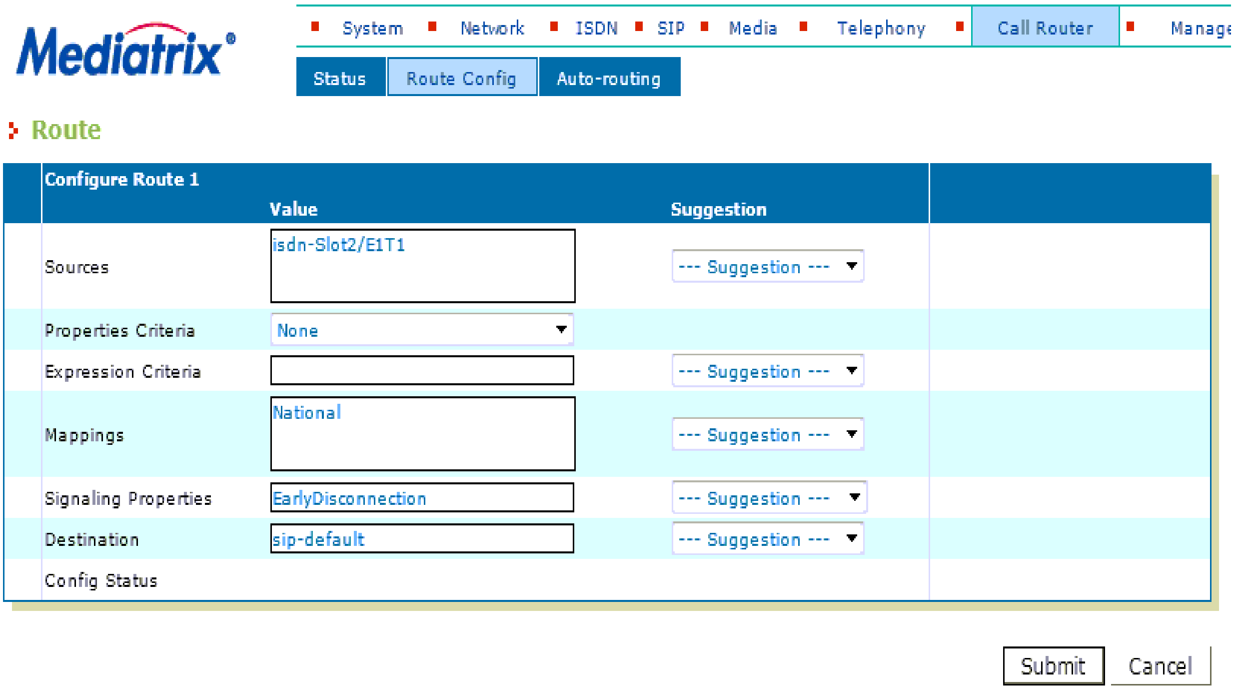
ISDN 30 seconds Timeout (Media Menu)

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:



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:

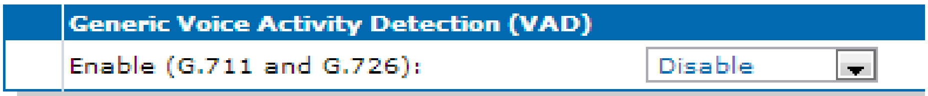


Media Settings (Media Menu)

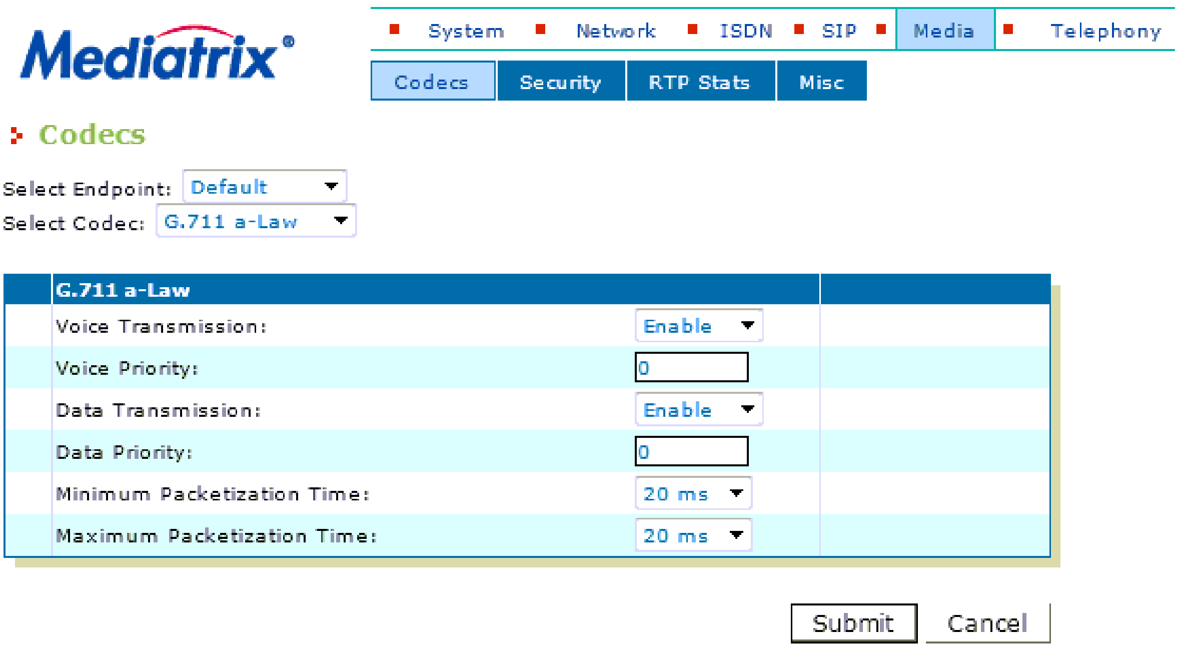
1. Configuring Codec (Media Menu – Codecs)

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

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

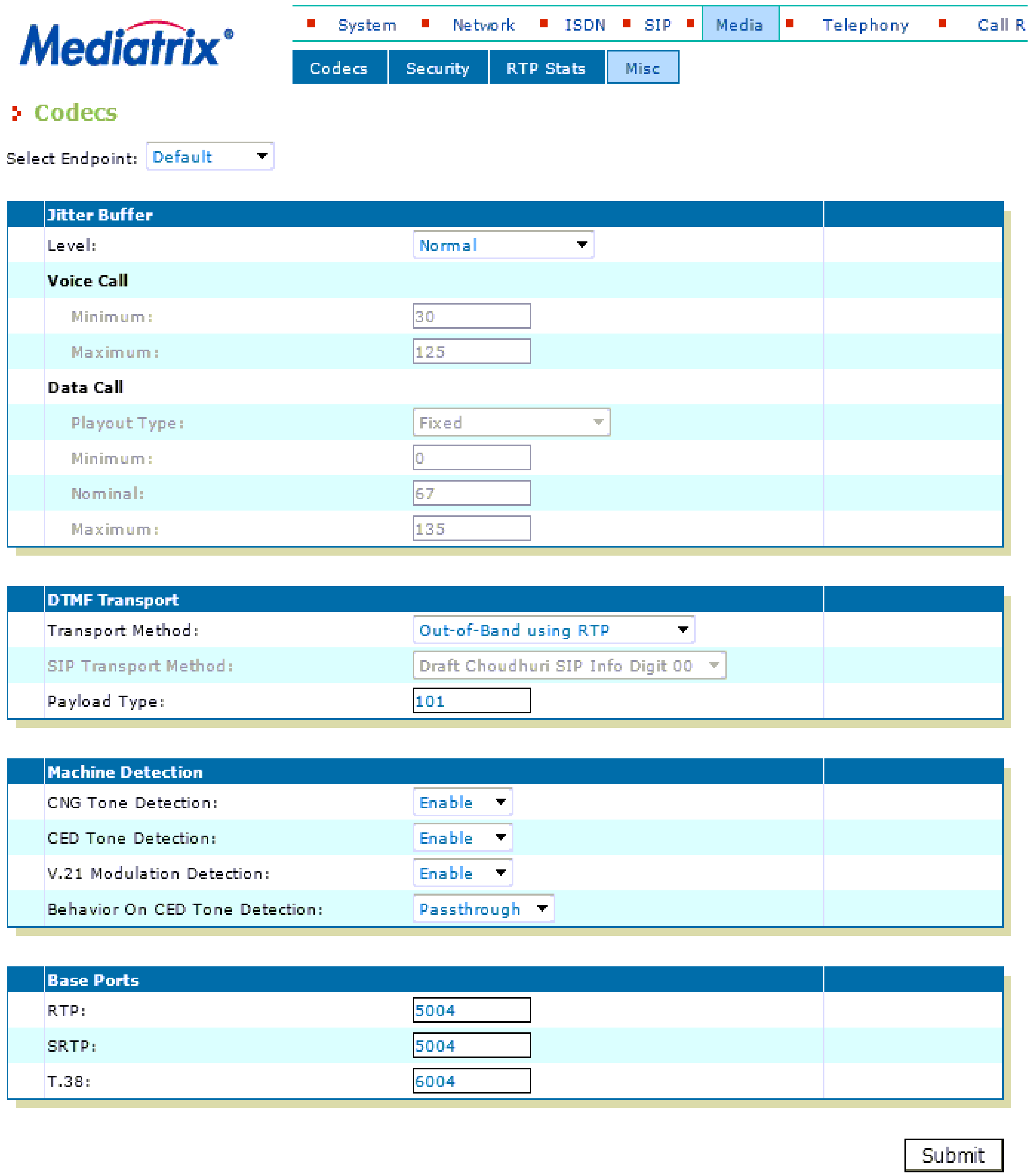


Click on Edit in the G711.a-Law window to open the advanced codec settings to set the packet length voice to 20ms as shown below:



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:



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.

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.