

# Mediatrix<sup>®</sup>

A Division of **media5** Corporation



## Configuration Notes 243

### Mediatrix 3000 Digital Gateway VoIP Trunking with a Legacy PBX

February 15, 2007

Proprietary

© 2007 Mediatrix Telecom, A Division of Media5 Corporation



**Table of Contents**

Table of Contents..... 2

Introduction ..... 3

Mediatrix 3000 Digital Gateway Overview ..... 3

Deployment Scenario..... 4

    Description ..... 4

    Objectives ..... 4

    Assumptions ..... 5

    Steps..... 5

Configuration of the Mediatrix 3000 DG Digital Gateway ..... 6

    Physical Connection of the Mediatrix 3000 DG to the Network and PBX..... 6

    IP Address Discovery or Configuration ..... 6

    Web Interface Access ..... 7

    SIP Configuration..... 8

    ISDN Configuration ..... 11

    Call Routing Configuration ..... 14

    Basic Call Establishment ..... 21

Further Information and Configuration ..... 22

Appendix A - Restarting a Service ..... 23



0243

## Introduction

This document outlines the configuration steps to set up a Mediatrix® 3000 digital gateway to provide VoIP trunking with a legacy PBX.

## Mediatrix 3000 Digital Gateway Overview

These configuration notes apply to the Mediatrix 3000 Series digital gateway products. The Mediatrix 3000 Series digital gateways allow enterprises to lower communications costs over any IP link. The platform supports ISDN E1 and T1 PRI telephony interfaces, as well as ISDN BRI interfaces. They provide an ideal solution for enterprise voice applications or for connecting to a service provider's broadband access.



Mediatrix® 3000 digital gateways are fully scalable in terms of number of ports and functionalities. They currently come in the following models:

Model	Interfaces	VoIP Call Capacity
Mediatrix 3404	5 BRI ports	up to 8
Mediatrix 3408	10 BRI ports	up to 16
Mediatrix 3531	1xT1-PRI interface	up to 23
Mediatrix 3532	2xT1-PRI interface	up to 46
Mediatrix 3631	1xE1-PRI interface	up to 30
Mediatrix 3632	2xE1-PRI interface	up to 60

The Mediatrix digital gateways link any standard ISDN E1/T1 PRI or BRI connection to the IP network and deliver the clarity of toll quality voice for a comprehensive VoIP solution.

T.38 FoIP, fax bypass, and modem bypass capabilities ensure that the Mediatrix digital gateways seamlessly transport voice and data services. The Mediatrix digital gateways offer flexibility and scalability for VoIP network integration and low bandwidth voice.

With configurable NT/TE PRI ports, call-switching and user programmable call routing (including caller/called ID), Mediatrix digital gateways integrate smoothly into existing PBX and PSTN networks.

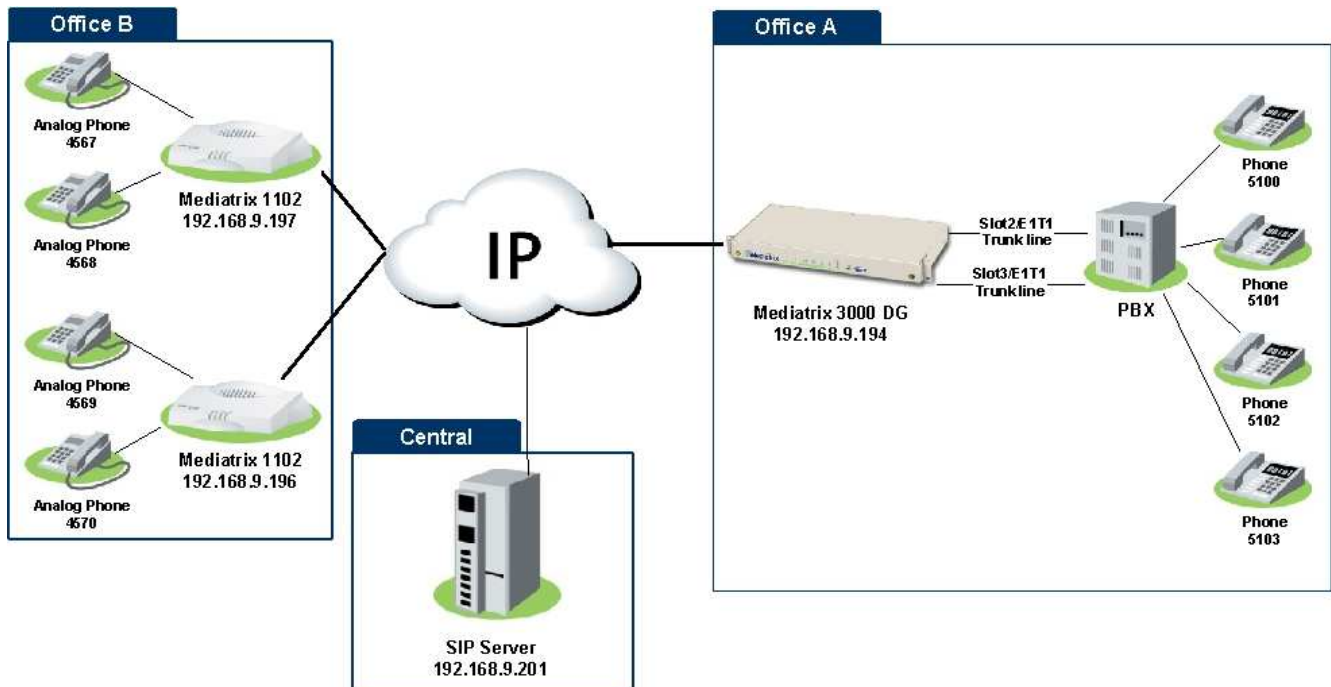
### Key Features:

- Voice Routing.
- Fax over IP support, including T.38.
- Proven voice algorithms implemented on dedicated DSP for enhanced voice quality.
- Up to 60 simultaneous calls.
- SNMPv3 and web management.
- Configuration file encryption.
- Automatic firmware and configuration file download.
- PSTN Bypass feature (BRI models only).

**Deployment Scenario**

**Description**

This configuration note is a step-by-step guide to set up one Mediatrix 3632 Series digital gateway to provide VoIP trunking with a legacy PBX. The Mediatrix 3632 is used to connect a branch office's PBX to an existing VoIP network. The configuration starts with the Mediatrix 3632 default configuration but can be easily customized for the 3631, 3531, and 3532, so from now on, the device will be referred to as the *Mediatrix 3000 DG* (Digital Gateway). The following is the network topology to which we will refer in our sample deployment.



**Figure 1 - Network Topology**

**Note:** The network addresses and phone numbers shown above are sample values that will most probably vary in your specific setup. In the following pages, when referring to such a sample value, it will be visually outlined (e.g., 192.168.9.194), so whenever you see parameters outlined in that fashion, you should replace them with the values that are appropriate for your specific setup.

**Objectives**

The steps described in the following pages will show you how to setup the Mediatrix 3000 DG so it can:

- A. receive calls from the PBX and route them to a remote branch through the VoIP network. (e.g., from Office A to Office B):
  1. a user from Office A picks up a phone and dials a number.
  2. the PBX uses one of its trunk lines to route the call to the Mediatrix 3000 DG.
  3. the Mediatrix 3000 DG forwards the call to the appropriate Mediatrix 1102.
  4. the Mediatrix 1102 makes the appropriate analog phone ring.
  5. a user in Office B picks up the analog phone and the call is established.
- B. receive calls from remote branches through the VoIP network and route them on one of the local branch's PBX lines (e.g., from Office B to Office A):
  1. a user from Office B picks up an analog phone and dials a number.
  2. the appropriate Mediatrix 1102 routes the call to the Mediatrix 3000 DG.



**0243**

3. the Mediatrix 3000 DG decides to which ISDN PRI interface route this call.
4. the Mediatrix 3000 DG routes the call to the PBX.
5. the PBX makes the appropriate phone ring.
6. a user from Office A picks up the phone and the call is established.

## Assumptions

This configuration note focuses on configuring the Mediatrix 3000 DG, and assumes that:

- extension numbers behind the PBX correspond to registered users in the SIP server (without authentication).
- the Mediatrix 3000 DG in Office A is connected to two E1 trunk lines in the PBX.
- the Office B setup is functional, and the SIP users are correctly registered to the SIP server.

## Steps

This configuration note will guide you through the following steps:

1. Physical connection of the Mediatrix 3000 DG to the network and PBX.
2. IP address discovery or configuration.
3. Web interface access.
4. SIP configuration.
5. ISDN configuration.
6. Call routing configuration.
7. Basic call establishment.



0243

## Configuration of the Mediatrix 3000 DG Digital Gateway

### Physical Connection of the Mediatrix 3000 DG to the Network and PBX

Please refer to the Mediatrix 3000 DG Quick Start booklet (packaged with the Mediatrix 3000 DG) for instructions on hardware installation.

The Mediatrix 3000 DG Quick Start booklet can also be found online on the Mediatrix Download Portal at <https://support.mediatrix.com/DownloadPlus/Download.asp>.

### IP Address Discovery or Configuration

*The purpose of this section is to be able to contact the Mediatrix 3000 DG's management interface to start with unit configuration.*

Once the physical connection is complete and the Mediatrix 3000 DG is powered up, the first thing to do is find out the IP address the Mediatrix 3000 DG is using. The Mediatrix 3000 DG's WAN IP address can be set either dynamically or statically. The default behaviour of the Mediatrix 3000 DG is to try to obtain a dynamic IP address through DHCP.

You can also access the Mediatrix 3000 DG through its private LAN interface.

#### Dynamic WAN IP Address Discovery

Before connecting the Mediatrix 3000 DG to the network, Mediatrix strongly suggests that you reserve an IP address in your DHCP server for the unit you are about to connect. DHCP servers reserve IP addresses for specific devices by using a unique identifier for each device. The Mediatrix 3000 DG's unique identifier is its media access control (MAC) address. The MAC address appears on the label located on the bottom side of the unit.

If you have not reserved an IP address, you can discover which IP address has been assigned to the Mediatrix 3000 DG by either:

- consulting your DHCP server's logs to find out details on the DHCP lease that was given to the Mediatrix 3000 DG.
- using a network packet sniffer (e.g., Ethereal) to examine the DHCP messages exchanged between the Mediatrix 3000 DG and your DHCP server while the Mediatrix 3000 DG boots up.

#### Default WAN Static IP Address Configuration

If there is no DHCP server in your network, then the WAN IP address can be configured statically. The first thing to do is set the Mediatrix 3000 DG to its known default static IP address. You can do this by using the Mediatrix 3000 DG's partial reset feature (see the section [Further Information and Configuration](#) for more details).

1. Once the Mediatrix 3000 DG has finished booting up (the Power LED is lit, not blinking), insert a small, unbent paper clip into the RESET/DEFAULT hole located at the rear of the Mediatrix 3000 DG and press the RESET/DEFAULT button. The Power LED will start blinking, and after a few seconds, all the LEDs will start blinking. Release the paper clip after all the LEDs start blinking and before they all stop blinking (between 7-11 seconds).

After a partial reset is performed, the Mediatrix 3000 DG's WAN connection uses the default 192.168.0.1 IP address. From now on, you can optionally change the Mediatrix 3000 DG's IP address (see section [Further Information and Configuration](#) for more details).

#### LAN Interface Access

The Mediatrix 3000 DG's default LAN IP address is 192.168.0.10.



0243

## Web Interface Access

The purpose of this section is to log in to the Mediatrix 3000 DG's web interface.

The Mediatrix 3000 DG's web interface may be used to view the status of the Mediatrix 3000 DG and set its numerous parameters.

1. In your web browser's address field, type **192.168.9.194** (or the address of the Mediatrix 3000 DG). The PC you use must be connected to the same subnet as the Mediatrix 3000 DG or to a network where it can reach the Mediatrix 3000 DG's IP address. The following window appears:

A screenshot of the Mediatrix web interface login page. At the top left is the Mediatrix logo. Below it is a green message: 'Please enter your username and password'. There are two input fields: 'User Name:' and 'Password:'. Below the 'Password:' field is a 'Login' button.

2. Enter the user name **public**. Leave the *Password* field empty.

A screenshot of the Mediatrix web interface login page, similar to the previous one. The 'User Name:' field now contains the text 'public', which is circled in red. The 'Password:' field is empty. The 'Login' button is still present.

3. Click **Login**.

A screenshot of the Mediatrix web interface main page. At the top left is the Mediatrix logo. To the right is a navigation menu with tabs: 'System', 'Network', 'ISDN', and 'SIP'. Below this is another set of tabs: 'Information', 'Services', and 'Syslog'. The 'Information' tab is selected. Below the tabs is a green message: 'Information'. There are two tables. The first table is titled 'Current Status' and contains the following data:

Current Status	
System Description:	Mediatrix 3301-001
Serial Number:	000340003P122060017
Firmware Version:	1.1.4.32
MAC Address:	0090f802b277
System Uptime (D:HH:MM:SS):	0:00:21:13
SNMP Port:	161

The second table is titled 'Installed Hardware' and contains the following data:

Name	Serial Number	Location
Mediatrix 3301-020	00046000126060015	Slot2

You now have access to the Mediatrix 3000 DG's configuration web interface.



### SIP Configuration

The purpose of this section is to setup the Mediatrix 3000 DG to use your SIP server for registration and call routing, and to tell the Mediatrix 3000 DG to register SIP users for all the phones that are connected to the PBX.

The SIP configuration tells the Mediatrix 3000 DG which SIP servers, parameters, and phone numbers to use. The following steps configure the Mediatrix 3000 DG as illustrated in the sample network topology.

1. Click the **SIP** menu, then the **Servers** sub-menu. The following window appears:

The screenshot shows the Mediatrix configuration interface. At the top, there are tabs for System, Network, ISDN, SIP, and Telephony. Under the SIP tab, there are sub-tabs for Gateways, Servers, Registrations, Endpoints, and Authentication. The 'Servers' sub-tab is selected. Below this, there are three main sections:

- SIP Default Servers:** A table with fields for Registrar Host (192.168.10.10:0), Proxy Host (192.168.10.10:0), and Outbound Proxy Host.
- SIP Gateway Specific Registrar Servers:** A table with columns for Gateway Name, Gateway Specific, and Registrar Host. The 'default' gateway has 'No' selected for Gateway Specific and 192.168.0.10:0 for Registrar Host.
- SIP Gateway Specific Proxy Servers:** A table with columns for Gateway Name, Gateway Specific, Proxy Host, and Outbound Proxy Host. The 'default' gateway has 'No' selected for Gateway Specific, 192.168.0.10:0 for Proxy Host, and 0.0.0.0:0 for Outbound Proxy Host.

At the bottom right, there are two buttons: 'Submit' and 'Submit & Refresh Registration'.

2. Set the *Registrar Host* field to the address of the central SIP Server **192.168.9.201**.
3. Set the *Proxy Host* field to the address of the central SIP Server **192.168.9.201**.

This screenshot is identical to the previous one, but the 'Registrar Host' and 'Proxy Host' fields in the 'SIP Default Servers' section are now set to '192.168.9.201'. These values are circled in red in the original image to highlight the changes.

4. Click **Submit** to save the configuration changes. The Mediatrix 3000 DG is now configured to use your SIP server.





5. Click the **Registrations** sub-menu. The following window appears:

System						Network						ISDN						SIP						Telephony						Management																																									
Gateways												Servers												Registrations												Endpoints												Authentication												Misc											
<b>Registrations</b>																																																																							
Endpoints Registration																																																																							
Endpoint	User Name	Friendly Name	Register	Gateway Name																																																																			
Slot2/E1T1	<input type="text"/>	<input type="text"/>	Disable	all																																																																			
Slot3/E1T1	<input type="text"/>	<input type="text"/>	Disable	all																																																																			
Unit Registration																																																																							
Index	User Name	Gateway Name																																																																					
																						+																																																	
Submit												Submit & Refresh Registration																																																											

In this window, you can enter the extension numbers from the PBX to be registered in the SIP server.

6. Click the **+** button at the bottom right of the *Unit Registration*, section. An empty entry appears in the section.

System						Network						ISDN						SIP						Telephony						Management																																									
Gateways												Servers												Registrations												Endpoints												Authentication												Misc											
<b>Registrations</b>																																																																							
Endpoints Registration																																																																							
Endpoint	User Name	Friendly Name	Register	Gateway Name																																																																			
Slot2/E1T1	<input type="text"/>	<input type="text"/>	Disable	all																																																																			
Slot3/E1T1	<input type="text"/>	<input type="text"/>	Disable	all																																																																			
Unit Registration																																																																							
Index	User Name	Gateway Name																																																																					
1	<input type="text"/>	all																																																																					
																						-																																																	
																						+																																																	
Submit												Submit & Refresh Registration																																																											

7. Enter the phone number of the first phone from Office A of the sample network topology ([Figure 1](#)) in the *User Name* field (**5100** in our example).

System						Network						ISDN						SIP						Telephony						Management																																									
Gateways												Servers												Registrations												Endpoints												Authentication												Misc											
<b>Registrations</b>																																																																							
Endpoints Registration																																																																							
Endpoint	User Name	Friendly Name	Register	Gateway Name																																																																			
Slot2/E1T1	<input type="text"/>	<input type="text"/>	Disable	all																																																																			
Slot3/E1T1	<input type="text"/>	<input type="text"/>	Disable	all																																																																			
Unit Registration																																																																							
Index	User Name	Gateway Name																																																																					
1	5100	all																																																																					
																						-																																																	
																						+																																																	
Submit												Submit & Refresh Registration																																																											



0243

- Repeat steps 6 and 7 for all of the phones from Office A of the sample network topology ([Figure 1](#)). At the end of the process, the *Unit Registration* section looks like the following:

The screenshot shows the Mediatrix configuration interface. At the top, there are navigation tabs: System, Network, ISDN, SIP, Telephony, and Management. Below these are sub-tabs: Gateways, Servers, Registrations, Endpoints, Authentication, and Misc. The 'Registrations' section is active and contains two tables.

**Endpoints Registration Table:**

Endpoint	User Name	Friendly Name	Register	Gateway Name
Slot2/E1T1	<input type="text"/>	<input type="text"/>	Disable	all
Slot3/E1T1	<input type="text"/>	<input type="text"/>	Disable	all

**Unit Registration Table:**

Index	User Name	Gateway Name	
1	<input type="text" value="S100"/>	all	-
2	<input type="text" value="S101"/>	all	-
3	<input type="text" value="S102"/>	all	-
4	<input type="text" value="S103"/>	all	-
			+

At the bottom of the interface, there are two buttons: 'Submit' and 'Submit & Refresh Registration'.

- Click **Submit & Refresh Registrations**. This saves the configuration in the Mediatrix 3000 DG and causes it to send the appropriate SIP REGISTER messages to the SIP server so each phone has a registered SIP user associated with it.
- OPTIONAL STEP: if your SIP server requires SIP authentication, further configuration steps are necessary so the Mediatrix 3000 DG has all the needed information to authenticate to the server (see the section [Further Information and Configuration](#) for more details).



0243

### ISDN Configuration

The purpose of this section is to configure the Mediatrix 3000 DG's ISDN PRI interfaces in Network mode (NT) for an E1 line type. This requires that the PBX be configured in Terminal Equipment (TE). If your setup differs, please refer to the section [Further Information and Configuration](#) for more details.

The ISDN configuration tells the Mediatrix 3000 DG how its ISDN PRI interfaces should behave. You must configure the ISDN parameters of the Mediatrix 3000 DG digital gateways for each interface you intend to use.

- 1. Click the **ISDN** menu, then the **Primary Rate Interface** sub-menu. The following window appears:

The screenshot shows the Mediatrix configuration interface. At the top, there are tabs for System, Network, ISDN, and SIP. Under the ISDN tab, there are sub-tabs for Status and Primary Rate Interface. The 'Primary Rate Interface' sub-tab is active. Below it, there is a 'Select Interface:' dropdown menu with 'Slot2/E1T1' selected. The main configuration area is divided into two sections: 'Hardware Configuration' and 'Interface Configuration'. The 'Hardware Configuration' section has a 'Clock Reference (Applies to the slot):' dropdown set to 'None'. The 'Interface Configuration' section contains various settings, all of which are currently set to 'Disable' or 'Enable' as follows:

Parameter	Value
Endpoint Type:	TE
Line Type:	E1
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:	0
Signal Information Element:	Disable
Inband Tone Generation:	Enable
Inband DTMF Dialing:	Enable
Overlap Dialing:	Enable
Calling Name Max Length:	34
Exclusive B-Channel Selection:	Disable
Sending Complete:	Enable
Calling Line Information Presentation:	Disable
Calling Line Information Restriction:	Disable
Calling Line Information Restriction Override:	Disable
Send Restart On Startup:	Enable

At the bottom right of the configuration area is a 'Submit' button.

- 2. Select the interface for which you want to apply the changes in the *Select Interface* drop-down menu. Depending on the model of Mediatrix 3000 DG you are using, you may have 1 or 2 interfaces available in the drop-down menu.

This close-up shows the 'Select Interface:' dropdown menu. The menu is open, showing three options: 'Slot2/E1T1', 'Slot2/E1T1', and 'Slot3/E1T1'. The first two options are highlighted with a red circle, indicating they are the available choices for configuration.



0243

- In the *Interface Configuration* section, set the *Endpoint Type* field to **NT**. Leave all other parameters to their default values.

NOTE: The *Line Coding*, *Line Framing*, and *Signaling Protocol* fields are left to common default values here. However, they must be compatible with your setup. If your setup differs, please refer to the section [Further Information and Configuration](#) for more details.

System Network ISDN SIP

Status Primary Rate Interface

Primary Rate Interface

Select Interface: Slot2/E1T1

Hardware Configuration	
Clock Reference (Applies to the slot):	None

Interface Configuration	
Endpoint Type:	NT
Line Type:	E1
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:	0
Signal Information Element:	Disable
Inband Tone Generation:	Enable
Inband DTMF Dialing:	Enable
Overlap Dialing:	Enable
Calling Name Max Length:	34
Exclusive B-Channel Selection:	Disable
Sending Complete:	Enable
Calling Line Information Presentation:	Disable
Calling Line Information Restriction:	Disable
Calling Line Information Restriction Override:	Disable
Send Restart On Startup:	Enable

Submit

- Click **Submit** to apply the configuration changes made to this interface.

System Network ISDN SIP

Status Primary Rate Interface

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

Primary Rate Interface

Select Interface: Slot2/E1T1

Hardware Configuration	
Clock Reference (Applies to the slot):	None

Interface Configuration	
Endpoint Type:	NT
Line Type:	E1



# Configuration Notes

0243

5. The parameters that have just been configured require a restart of the ISDN service. A service is a logical grouping of features. Restarting a service is a required mechanism for certain elements in the configuration. However, you can finish with the ISDN configuration steps before doing that. Once the ISDN configuration is over, follow the instructions from [Appendix A - Restarting a Service](#) to restart the ISDN service as required.
6. Repeat steps 2-3-4 for all of the ISDN PRI interfaces listed in the *Select Interface* field.
7. Restart the ISDN service as described in [Appendix A - Restarting a Service](#).
8. To confirm that the ISDN configuration is completed and compatible with the rest of your setup, click the **Status** sub-menu. The following window appears:

The screenshot shows the Mediatrix configuration interface. At the top, there are tabs for System, Network, ISDN, and SIP. Below these is a sub-menu for Status, with 'Primary Rate Interface' selected. The main content area is titled 'Status' and contains two sections for ISDN PRI interfaces: Slot2/E1T1 and Slot3/E1T1. Each section has a 'Configure' link. In both sections, 'Physical Link' and 'Signaling' are shown as 'Up', with these labels circled in red. Below each section is a 'Bearer Group' table with 30 channels, all of which are shown as 'Free'.

Location	Clock Reference (Applies to the slot)
Slot2:	None
Slot3:	Other Card

Slot2/E1T1 [Configure]			
Physical Link: Up			
Signaling: Up			
Bearer Group			
Channel 1: Free	Channel 9: Free	Channel 17: Free	Channel 25: Free
Channel 2: Free	Channel 10: Free	Channel 18: Free	Channel 26: Free
Channel 3: Free	Channel 11: Free	Channel 19: Free	Channel 27: Free
Channel 4: Free	Channel 12: Free	Channel 20: Free	Channel 28: Free
Channel 5: Free	Channel 13: Free	Channel 21: Free	Channel 29: Free
Channel 6: Free	Channel 14: Free	Channel 22: Free	Channel 30: Free
Channel 7: Free	Channel 15: Free	Channel 23: Free	
Channel 8: Free	Channel 16: Free	Channel 24: Free	

Slot3/E1T1 [Configure]			
Physical Link: Up			
Signaling: Up			
Bearer Group			
Channel 1: Free	Channel 9: Free	Channel 17: Free	Channel 25: Free
Channel 2: Free	Channel 10: Free	Channel 18: Free	Channel 26: Free
Channel 3: Free	Channel 11: Free	Channel 19: Free	Channel 27: Free
Channel 4: Free	Channel 12: Free	Channel 20: Free	Channel 28: Free
Channel 5: Free	Channel 13: Free	Channel 21: Free	Channel 29: Free
Channel 6: Free	Channel 14: Free	Channel 22: Free	Channel 30: Free
Channel 7: Free	Channel 15: Free	Channel 23: Free	
Channel 8: Free	Channel 16: Free	Channel 24: Free	

The *Physical Link* and *Signaling* status fields of each of the ISDN PRI interfaces configured should be **Up**. If they are not, please review the configuration steps from this section of the document and make sure they have been applied correctly and are compatible with your setup. Please refer to the section [Error! Reference source not found.](#) for more details.



## Call Routing Configuration

The purpose of this section is to configure the Mediatrix 3000 DG's call router so it can route calls to/from the VoIP network and the PBX as described in the [Deployment Scenario](#) section.

You must configure the call router parameters of the Mediatrix 3000 DG digital gateway so that the calls can properly terminate. Remember that the purpose of this configuration note is to achieve the sample deployment scenario shown in [Figure 1](#). Your specific setup may vary.

### Planning the Call Router

The goal of planning the Call router configuration is to summarize the rules incoming calls will follow when passing through the Mediatrix 3000 DG.

This is:

- Call sources and destinations.
- Calls allowed and rejected.
- Call properties manipulations.
- All routing possibilities.

Before going further with the configuration steps, you should refer back to the two types of calls described in the [Error! Reference source not found.](#) section.

The most basic call scenario implies at least configuring *Routes*. In the current deployment scenario, you will also configure a *Hunt Group* to support step 3 of call scenario B defined in the [Error! Reference source not found.](#) section (see [Further Information and Configuration](#) for more details).

- A Route is a virtual connection made inside the Mediatrix 3000 DG between call sources and destinations. Routes are part of the Mediatrix 3000 DG's Route table. When a call comes in, the Mediatrix 3000 DG uses its Route table to decide to which destination route the call.
- A Hunt Group is a virtual entity that regroups different call destinations in one group. This entity can then be used as a call destination in a Route. When an incoming call is routed to a Hunt, the Hunt uses an algorithm to decide which one of its internal destinations the call is effectively routed to.



**Configuring the Call Router**

**Hunt Group**

*The purpose of this subsection is to configure a Hunt Group in the Mediatrix 3000 DG, so it can be later used as a route's destination.*

In the current scenario, you will use a Hunt Group to group both of the Mediatrix 3632's ISDN PRI interfaces as one virtual call destination.

1. Click the **Telephony** menu, then the **Call Routing Config** sub-menu. The following window appears.

**Mediatrix**  
A Division of **mediala** Corporation

System ■ Network ■ ISDN ■ SIP ■ **Telephony** ■ Management

DTMF Maps CODECS Call Routing Status **Call Routing Config** Misc

➤ **Call Routing Config**

Config Modified: no

Route									
Index	Source	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions		
+									

Mapping Type				
Index	Name	Criteria	Transformation	Actions
+				

Mapping Expression						
Index	Name	Criteria	Transformation	Sub Mappings	Actions	
+						

Signaling Properties										
Index	Name	Early Connect	Early Disconnect	Destination Host	Allow 180 with SDP	Allow 183 without SDP	Privacy	SIP Headers Translations	Call Properties Translations	Actions
+										

SIP Headers Translations					
Index	Name	SIP Header	Built From	Fix Value	Actions
+					

Call Properties Translations					
Index	Name	Call Property	Built From	Fix Value	Actions
+					

Hunt						
Index	Name	Destinations	Selection Algorithm	Timeout (seconds)	Causes	Actions
+						

Apply Rollback

2. Locate the *Hunt* section at the bottom of the window.



- Click the **+** button at the bottom right of the *Hunt* section. The following window appears.

To create a Hunt Group:

- Set the *Name* field to **hunt\_PBX**.
- Use the *Suggestion* drop-down list to select and add the possible destinations that will be part of the Hunt Group.

- Following the [Deployment Scenario](#), select one by one both of the Mediatrix 3000 DG's ISDN PRI interfaces (**isdn-Slot2/E1T1**, which corresponds to port E1/T1 on SLOT 2 and **isdn-Slot3/E1T1**, which corresponds to port E1/T1 on SLOT 3). The interfaces will be automatically added as destinations for that Hunt Group.





**Configuration Notes**

**0243**

7. Leave the other fields with their default value.

**Configure Hunt End**

Name	Value	Suggestion
Name	hunt_PBX	
Destinations	isdn-Slot2/E1T1, isdn-Slot3/E1T1	--- Suggestion ---
Selection Algorithm	Sequential	
Timeout (seconds)	0	
Causes	31, 34, 38, 41, 42, 43, 44, 47	--- Suggestion ---
Config Status		

Submit Cancel

8. Verify if the ISDN interfaces have been successfully added to the configuration by checking the *Destinations* field, then click **Submit** to apply changes and save the new Hunt Group.

**Call Routing Config**

Config Modified: yes

Route	Index	Source	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions
								+

Mapping Type	Index	Name	Criteria	Transformation	Actions
					+

Mapping Expression	Index	Name	Criteria	Transformation	Sub Mappings	Actions
						+

Signaling Properties	Index	Name	Early Connect	Early Disconnect	Destination Host	Allow 180 with SDP	Allow 183 without SDP	Privacy	SIP Headers Translations	Call Properties Translations	Actions
											+

SIP Headers Translations	Index	Name	SIP Header	Built From	Fix Value	Actions
						+

Call Properties Translations	Index	Name	Call Property	Built From	Fix Value	Actions
						+

Hunt	Index	Name	Destinations	Selection Algorithm	Timeout (seconds)	Causes	Actions
	1	hunt_PBX	isdn-Slot2/E1T1, isdn-Slot3/E1T1	Sequential	0	31, 34, 38, 41, 42, 43, 44, 47	Edit + -

Apply Rollback




9. You are brought back to the **Call Routing Config** sub-menu, and you can see the *Hunt Group* you have just created in the *Hunt* section.

You can also see a yellow Yes that warns you that the configuration has been modified but not applied (i.e., the **Call Routing Status** differs from the **Call Routing Config**). The *Call Routing Config* sub-menu is a working area where you build up a Call Router configuration. While you work in this area, the configured parameters are saved but not applied (i.e., they are not used to process incoming calls). The yellow Yes flag warns you that the configuration has been modified but is not applied. You will apply the configuration later when it is complete.

**Route**

The purpose of this subsection is to configure the Mediatrix 3000 DG so it makes virtual “connections” between call sources and destinations.

1. Locate the *Route* section at the top of the window.
2. Click the  button at the bottom right of the *Route* section. The following window appears.

Configure Route End	Value	Suggestion
Source	<input type="text"/>	--- Suggestion ---
Properties Criteria	None	
Expression Criteria	<input type="text"/>	--- Suggestion ---
Mappings	<input type="text"/>	--- Suggestion ---
Signaling Properties	<input type="text"/>	--- Suggestion ---
Destination	<input type="text"/>	--- Suggestion ---
Config Status		

Submit Cancel

3. To create a route from SIP (**sip-default**) to ISDN (**hunt\_PBX**), set the *Source* field to **sip-default** and the *Destination* field to **hunt-hunt\_PBX**. You can use both fields' associated *Suggestion* drop-down list to help you fill them. This route will satisfy call scenario B described in section [Deployment Scenario](#), where SIP users from Office B call phones from Office A.

Configure Route End	Value	Suggestion
Source	sip-default	--- Suggestion ---
Properties Criteria	None	
Expression Criteria	<input type="text"/>	--- Suggestion ---
Mappings	<input type="text"/>	--- Suggestion ---
Signaling Properties	<input type="text"/>	--- Suggestion ---
Destination	hunt-hunt_PBX	--- Suggestion --- isdn-Slot2/E1T1 isdn-Slot3/E1T1 sip-default <b>hunt-hunt_PBX</b> route-hunt-
Config Status		

Submit Cancel



4. Click **Submit** to apply changes and save the new route.

The screenshot shows the Mediatrix web interface for 'Call Routing Config'. At the top, there are navigation tabs for System, Network, ISDN, SIP, Telephony, and Management. Under 'Telephony', there are sub-tabs for DTMF Maps, CODECS, Call Routing Status, Call Routing Config, and Misc. A green arrow points to the 'Call Routing Config' sub-tab. Below this, a 'Config Modified:' status bar shows 'yes' in a yellow box. The main section is titled 'Route' and contains a table with the following data:

Index	Source	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions
1	sip-default	None				hunt-hunt_PBX	1-dit, +, -

Below the route table are several empty configuration sections, each with a '+' button in the Actions column:

- Mapping Type (Index, Name, Criteria, Transformation, Actions)
- Mapping Expression (Index, Name, Criteria, Transformation, Sub Mappings, Actions)
- Signaling Properties (Index, Name, Early Connect, Early Disconnect, Destination Host, Allow 180 with SDP, Allow 183 without SDP, Privacy, SIP Headers Translations, Call Properties Translations, Actions)
- SIP Headers Translations (Index, Name, SIP Header, Built From, Fix Value, Actions)
- Call Properties Translations (Index, Name, Call Property, Built From, Fix Value, Actions)
- Hunt (Index, Name, Destinations, Selection Algorithm, Timeout (seconds), Causes, Actions)

At the bottom right of the interface, there are 'Apply' and 'Rollback' buttons.

- You are brought back to the **Call Routing Config** sub-menu, and you can see the route you just created in the *Route* section. You can also see the yellow Yes that warns you that the configuration has been modified but not applied (i.e., the **Call Routing Status** differs from the **Call Routing Config**).
- Repeat steps 2 to 5 twice to create two additional routes. These new routes will satisfy call scenario A described in [Error! Reference source not found.](#), where phones from Office A call SIP users from Office B:
  - one from Source **isdn-Slot2/E1T1** to Destination **sip-default**, and
  - one from Source **isdn-Slot3/E1T1** to Destination **sip-default**.



0243

- After completing all the route configuration steps, you will see your three routes.

System  Network  ISDN  SIP  Telephony  Management

DTMF Maps CODECS Call Routing Status Call Routing Config Misc

**Call Routing Config**

Config Modified: **yes**

Route Index	Source	Properties	Criteria	Expression Criteria	Mappings	Signaling Properties	Destination	Actions
1	sip-default	None					hunt-hunt_PBX	Edit ↓ + -
2	isdn-Slot2/E1T1	None					sip-default	Edit ↑ + -
3	isdn-Slot3/E1T1	None					sip-default	Edit ↑ + -

Mapping Type Index	Name	Criteria	Transformation	Actions
				+

Mapping Expression Index	Name	Criteria	Transformation	Sub Mappings	Actions
					+

Signaling Properties Index	Name	Early Connect	Early Disconnect	Destination Host	Allow 180 with SDP	Allow 183 without SDP	Privacy	SIP Headers Translations	Call Properties Translations	Actions
										+

SIP Headers Translations Index	Name	SIP Header	Built From	Fix Value	Actions
					+

Call Properties Translations Index	Name	Call Property	Built From	Fix Value	Actions
					+

Hunt Index	Name	Destinations	Selection Algorithm	Timeout (seconds)	Causes	Actions
1	hunt_PBX	isdn-Slot2/E1T1, isdn-Slot3/E1T1	Sequential	0	31, 34, 38, 41, 42, 43, 44, 47	Edit + -

- Click **Apply**. This applies all the parameters from **Call Routing Config** to the system. You can also see that the yellow *Config Modified* **yes** flag is cleared.



9. The call routing parameters can be seen in the **Call Routing Status** window.

The screenshot shows the Mediatrix configuration interface. At the top, there are tabs for System, Network, ISDN, SIP, Telephony, and Management. Under the Telephony tab, there are sub-tabs for DTMF Maps, CODECS, Call Routing Status, Call Routing Config, and Misc. The 'Call Routing Status' sub-tab is active.

Below the sub-tabs, there is a 'Call Routing Status' section with a 'Config Modified' field set to 'no'.

The main configuration area contains several tables:

Route	Source	Properties Criteria	Expression Criteria	Mappings	Signaling Properties	Destination
	sip-default	None				hunt-hunt_PBX
	isdn-Slot2/E1T1	None				sip-default
	isdn-Slot3/E1T1	None				sip-default

Signaling Properties								
Name	Early Connect	Early Disconnect	Destination Host	Allow 180 with SDP	Allow 183 without SDP	Privacy	SIP Headers Translations	Call Properties Translations

SIP Headers Translations				
Index	Name	SIP Header	Built From	Fix Value

Call Properties Translations				
Index	Name	Call Property	Built From	Fix Value

Hunt				
Name	Destinations	Selection Algorithm	Timeout (seconds)	Causes
hunt_PBX	isdn-Slot2/E1T1, isdn-Slot3/E1T1	Sequential	0	31, 34, 38, 41, 42, 43, 44, 47

Available Interface (ISDN endpoints and SIP Gateways)	
Name	
isdn-Slot2/E1T1	
isdn-Slot3/E1T1	
sip-default	

The configuration note has prepared the system to perform calls in both directions.

### Basic Call Establishment

Once this configuration procedure is completed, you are ready to start making basic calls through your new Mediatrix 3000 DG, considering that the rest of your network's setup is configured properly.

#### Perform Basic Call (Scenario A)

- Pickup the phone that has the phone number **5100**.
- Dial **4567**.
- The analog phone number **4567** rings.
- Pick up the analog phone number **4567**.
- The call is established.
- Hang up both phones to end the call.

#### Perform Basic Call (Scenario B)

- Pick up the analog phone number **4567**.
- Dial **5100**.
- The phone number **5100** rings.
- Pickup the phone number **5100**.



0243

- The call is established.
- Hang up both phones to end the call.

## Further Information and Configuration

You can refer to the following documents/sections for further information on configuration parameters and features used in this configuration note.

All documents are available online on the Mediatrix Download Portal at <https://support.mediatrix.com/DownloadPlus/Download.asp>.

- 1- For more information on the Partial Reset feature, and on what to do after performing a Partial Reset to recover a unit with which you had lost contact, refer to the *Partial Reset* section of the *Mediatrix 3000 Series Digital Gateway Software Configuration Guide*.
- 2- For more information on configuring level 2 network links, level 3 network interfaces and IP addresses, refer to the *Interfaces Configuration* section of the *Mediatrix 3000 Series Digital Gateway Software Configuration Guide*.
- 3- For more information on configuring the Mediatrix 3000 DG's ISDN PRI interfaces in TE or NT mode and additional parameters, refer to the *ISDN Configuration* section of the *Mediatrix 3000 Series Digital Gateway Software Configuration Guide*.
- 4- For more information on configuring the Mediatrix 3000 DG to work with SIP servers that require SIP authentication, refer to the *SIP Authentication* section of the *Mediatrix 3000 Series Digital Gateway Software Configuration Guide*.
- 5- For information on how to configure the Mediatrix 3000 DG so it processes dialed DTMFs according to specific dialing plans, refer to the *DTMF Maps Configuration* section of the *Mediatrix 3000 Series Digital Gateway Software Configuration Guide*.
- 6- For more information on call routing including routes, mappings, signaling properties, and hunts, refer to the *Call Router Configuration* section of the *Mediatrix 3000 Series Digital Gateway Software Configuration Guide*.



## Appendix A - Restarting a Service

The Mediatrix 3000 DG's features are divided in logical entities called *Services*. Some parameters in the Mediatrix 3000 DG require that the service to which they belong be restarted when they are configured in order for their new configuration value to be correctly applied. When this happens (usually after you click a **Submit** button), a message and a **Services** link are displayed at the top of the window stating that a service must be restarted.

In this example, a parameter of the ISDN services requires that this service be restarted.

1. Click the **Services** link, which brings you to the **Services** page. In this page, each service that requires to be restarted has a "\*" besides its name, as illustrated in the following window.

Service	Class	Status	Action	Comment
Authentication, Authorization and Accounting (AAA):	System	Started	[Dropdown]	
Basic Network Interface (BNI):	User	Started	[Dropdown]	
Call Routing (CROUT):	User	Started	[Dropdown]	
Certificate Manager (CERT):	System	Started	[Dropdown]	
Configuration Manager (CONF):	System	Started	[Dropdown]	
Device Control Manager (DCM):	System	Started	[Dropdown]	
Endpoint Administration (EPADM):	User	Started	[Dropdown]	
Endpoint Services (EPSERV):	User	Started	[Dropdown]	
Ethernet Manager (ETH):	System	Started	[Dropdown]	
Firmware Pack Updater (FPU):	System	Started	[Dropdown]	
Host Configuration (HOC):	System	Started	[Dropdown]	
* Integrated Services Digital Network (ISDN):	User	Started	[Dropdown]	
Local Quality Of Service (LQOS):	System	Started	[Dropdown]	
Media IP Transport (MIPT):	User	Started	[Dropdown]	
Notifications and Logging Manager (NLM):	User	Started	[Dropdown]	
Process Control Manager (PCM):	System	Started	[Dropdown]	
Service Controller Manager (SCM):	System	Started	[Dropdown]	
SIP Endpoint (SIPEP):	User	Started	[Dropdown]	
Simple Network Management Protocol (SNMP):	User	Started	[Dropdown]	
Telephony Interface (TELIF):	User	Started	[Dropdown]	
Web (WEB):	User	Started	[Dropdown]	



0243

- Restart each service that has a "\*" besides its name by clicking the **Restart** action so it correctly applies its new configuration.

* Integrated Services Digital Network (ISDN):	User	Started	<input type="button" value="Restart"/>
Local Quality Of Service (LQOS):	System	Started	<input type="button" value="Restart"/>
Media IP Transport (MIPT):	User	Started	<input type="button" value="Restart"/>

- Restarting a service may require other services to be restarted. This is why you would see a few services go from the stopping to starting to started states, even if you only restarted one service. The displayed status may be refreshed at any time by clicking the **Services** submenu or the **here** link.

Note: A "\*" beside the service name indicates that the service must be restarted to apply new configuration.

Services

Successfully sent the restart command to the service. Service statuses may have changed while the current page was loading, please click [here](#) to get the latest statuses.

Service	Class	Status	Action	Comment
Authentication, Authorization and Accounting (AAA):	System	Started	<input type="button" value="Restart"/>	
Basic Network Interface (BNI):	User	Started	<input type="button" value="Restart"/>	
Call Routing (CROUT):	User	Stopping	<input type="button" value="Restart"/>	
Certificate Manager (CERT):	System	Started	<input type="button" value="Restart"/>	
Configuration Manager (CONF):	System	Started	<input type="button" value="Restart"/>	
Device Control Manager (DCM):	System	Started	<input type="button" value="Restart"/>	
Endpoint Administration (EPADM):	User	Stopping	<input type="button" value="Restart"/>	
Endpoint Services (EPSERV):	User	Stopping	<input type="button" value="Restart"/>	
Ethernet Manager (ETH):	System	Started	<input type="button" value="Restart"/>	
Firmware Pack Updater (FPU):	System	Started	<input type="button" value="Restart"/>	
Host Configuration (HOC):	System	Started	<input type="button" value="Restart"/>	
Integrated Services Digital Network (ISDN):	User	Stopping	<input type="button" value="Restart"/>	
Local Quality Of Service (LQOS):	System	Started	<input type="button" value="Restart"/>	
Media IP Transport (MIPT):	User	Stopping	<input type="button" value="Restart"/>	
Notifications and Logging Manager (NLM):	User	Started	<input type="button" value="Restart"/>	
Process Control Manager (PCM):	System	Started	<input type="button" value="Restart"/>	
Service Controller Manager (SCM):	System	Started	<input type="button" value="Restart"/>	
SIP Endpoint (SIPEP):	User	Stopping	<input type="button" value="Restart"/>	
Simple Network Management Protocol (SNMP):	User	Started	<input type="button" value="Restart"/>	
Telephony Interface (TELIF):	User	Stopping	<input type="button" value="Restart"/>	
Web (WEB):	User	Started	<input type="button" value="Restart"/>	

Thank you for using Mediatrix solutions!