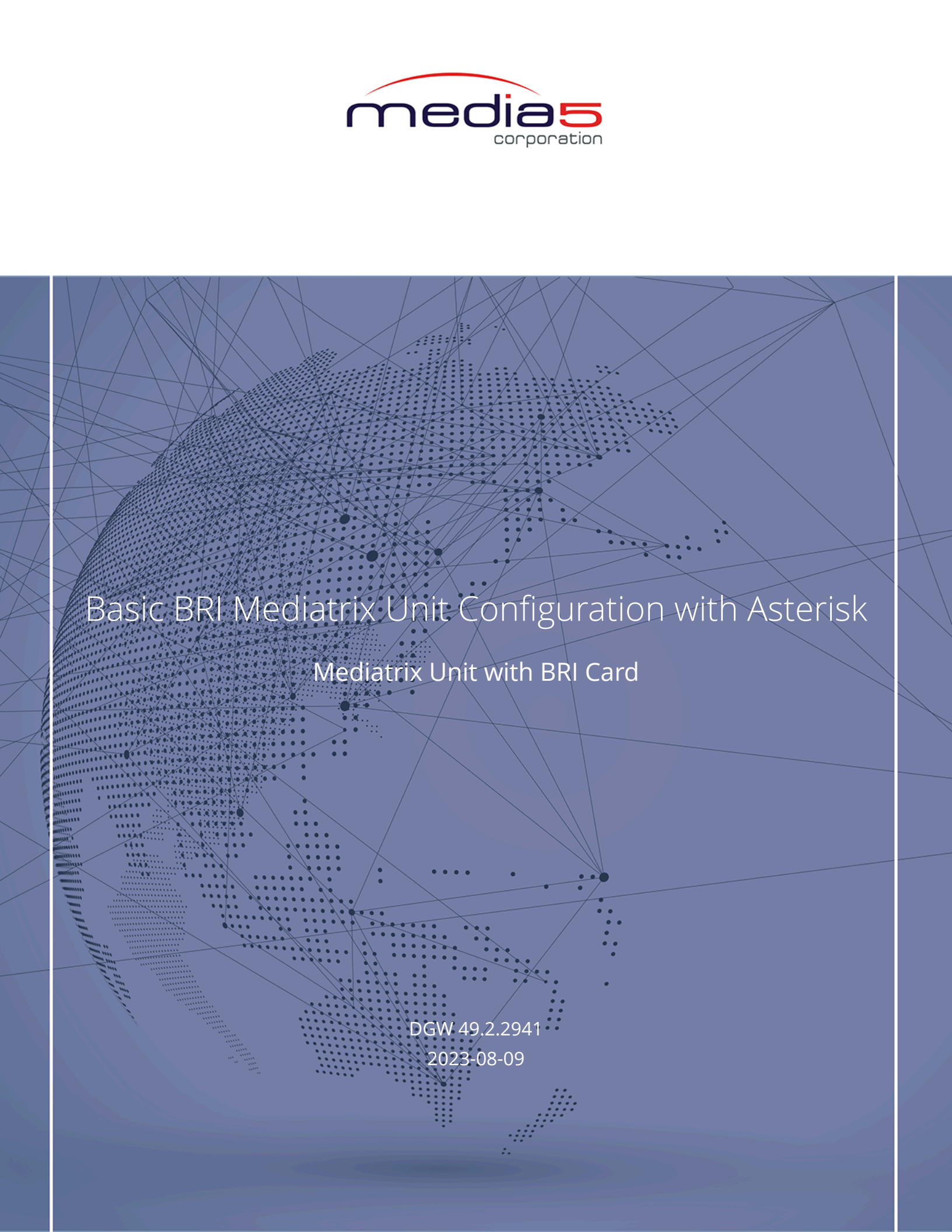




Basic BRI Mediatix Unit Configuration with Asterisk

Mediatix Unit with BRI Card

DGW 49.2.2941

2023-08-09

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1 Mediatix unit with Asterisk

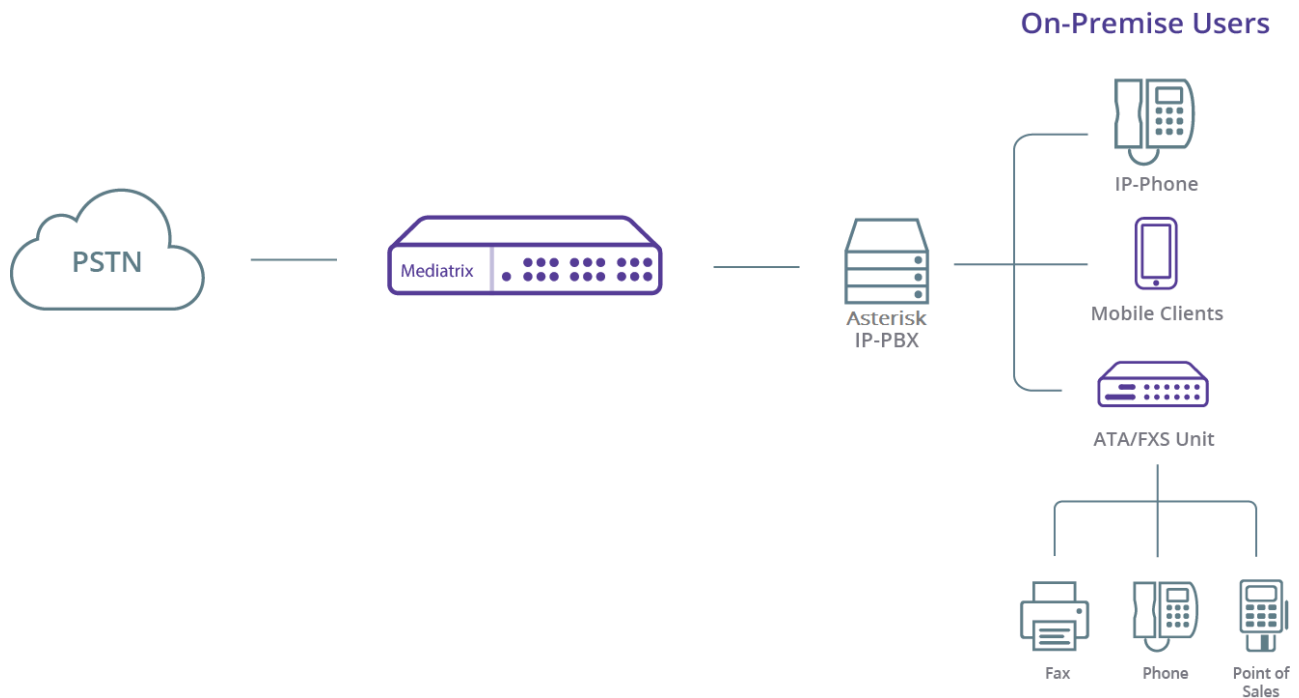
This document outlines the configuration steps required to connect a Mediatix unit to an Asterisk open-source telephone system.

In this scenario, the Mediatix unit is used to:

- Interface a PBX with an IP-PBX
- Provide PSTN access via analog lines
- Provide IP connectivity to analog phones and fax machines

The Asterisk IP-PBX provides:

- Call routing, Dial Plan
 - Including routes to local PSTN gateways
- Telephony services (voicemail, call forwarding, etc.) to IP users
- SIP Endpoints management
- Auto-Attendant



2 Requirements

- Asterisk server properly installed.
- SIP client properly configured and registered against the Asterisk IP-PBX, able to place and receive calls.

3 Information to Know Before Starting

Before starting to use these configuration notes, complete the following table to make sure you have the required information to complete the different steps.

IMPORTANT: If you are not familiar with the meaning of the fields and buttons, click **Show Help**, located at the upper right corner of the Web page. When activated, the fields and buttons that offer online help will change to green and if you hover over them, the description will be displayed.

Note: The Mediatrix unit must be reinitialised to its factory default settings to make sure the configuration can be successfully executed.

Information	Value	Used in Step
IP address of unit		Configuring the Asterisk - PBX Trunk (p.6) and Configuring the Asterisk - PSTN Lines (p.7)
Listening port of unit		Configuring the Asterisk - PBX Trunk (p.6) and Configuring the Asterisk - PSTN Lines (p.7)
The SIP username used for calls coming from the PSTN		Configuring the Asterisk - PSTN Lines (p.7)
The SIP username used for calls coming from the PBX		Configuring the Asterisk - PSTN Lines (p.7)
SIP Username and Password to authenticate the gateway		Authenticating the SIP Default Gateway (p.10)
Asterisk server IP address		
Asterisk server SIP listening port		
DTMF transport method		Configuring the Asterisk - PBX Trunk (p.6) and Configuring the Asterisk - PSTN Lines (p.7) and Configuring DTMF Transport (p.15)

4 Configuration of the PBX Trunk

4.1 Configuring the Asterisk - PBX Trunk

Steps

- 1) In the `sip.conf` configuration file, create a new extension and add:
 - a) [PBXTRUNK] : The SIP username used for calls coming from the PBX.
 - b) type=peer
 - c) host= IP address of Mediatix unit
 - d) port= listening port of the Mediatix unit
 - e) nat=no
 - f) qualify=no
 - g) canreinvite=no
 - h) dtmfmode=info
 - i) context=FromPBX
 - j) secret=TrunkPassword
 - k) t38pt_udptl=yes

Note: For more information on extensions, refer to the [Extension Description](#) (p.8) section.

- 2) In the `extensions.conf` file, add a context for calls coming from the PBX. Refer to your Asterisk documentation.
- 3) In the `extensions.conf` file, modify the context of the extensions to allow them to send calls to the PBX. Refer to your Asterisk documentation.
- 4) Reload the Asterisk settings by connecting to the Asterisk CLI (`asterisk -r`) and typing the `reload` command.
- 5) If you want to use the T.38 protocol to transfer faxes, refer to the [Technical Bulletin - Standard Fax Configuration](#) published on the [Media5 Documentation Portal](#).

Next Step

[Configuring the Asterisk - PSTN Lines \(p.7\)](#)

4.2 Configuring the Asterisk - PSTN Lines

Steps

- 1) In the `sip.conf` configuration file, create a new extension by adding the following:
 - a) `[PSTNTrunk]` = The SIP username used for calls coming from the PSTN.
 - b) `type= peer`
 - c) `host=IP address of Mediatrix unit`
 - d) `port=listening port of the Mediatrix unit`
 - e) `nat=no`
 - f) `qualify=no`
 - g) `canreinvite=no`
 - h) `dtmfmode=info`
 - i) `context= FromPSTN`
 - j) `secret= TrunkPassword`
 - k) `t38pt_udptl=yes`

Note: For more information on extensions, refer to the [Extension Description \(p.8\)](#) section.

- 2) In the `extensions.conf` file, add a context for calls coming from the PSTN. Refer to your Asterisk documentation.
- 3) In the `extensions.conf` file, modify the context of the PBX to allow the PBX users to send calls to the PSTN. Refer to your Asterisk documentation.
- 4) In the `extensions.conf`, modify the context of the extensions to allow them to send calls to the PBX. Refer to your Asterisk documentation.
- 5) You can reload the Asterisk settings by connecting to the Asterisk CLI (`asterisk -r`) and typing the `reload` command.

Next Step

[Configuring the Default SIP Servers \(p.8\)](#)

4.2.1 Extension Description

Extension	Description
[PBXTRUNK]	SIP username used for calls coming from the PBX.
[PSTNTrunk]	SIP username used for calls coming from the PSTN.
type=peer	This is part of the method used by the Asterisk server to match incoming INVITES to this user.
host=ip address of Mediatix unit	This means the extension will not register to the Asterisk server. This is the IP address of the Mediatix unit.
port=xxx	Port used for requests to and from the Mediatix unit.
nat=no	The Mediatix unit is not behind a NAT.
qualify=no	No keep alive is used.
canreinvite=no	No reINVITE is sent to this extension.
dtmfmode=info	The DTMF is sent/received in SIP INFO messages.
context=xxxx	This is the context where the call from this extension is sent. It is the same context for the IP phone and the Mediatix Unit.
secret=TrunkPassword	The SIP authentication password
t38pt_udptl=yes	This allows T.38 fax to be sent by this trunk. This setting is set to no for the IP phone and the Mediatix unit .

4.3 Configuring the Default SIP Servers

Information

In some instances, the configuration of the default servers may already be completed.

Steps

- 1) Go to **SIP/Servers**.
- 2) In the **Registrar Host** field, indicate the server IP address or FQDN of the SIP Registrar server .

- 3) In the **Proxy Host** field, indicate the server IP address or FQDN of the SIP Proxy server.
- 4) In the **Messaging Server Host** field, indicate the server IP address or FQDN of the Messaging server, if needed.
- 5) Leave the **Outbound Proxy Host** field empty, unless specifically instructed to set a value.

Note: Setting the address to 0.0.0.0:0 or leaving the field empty disables the outbound proxy host.

- 6) Click **Apply**.
- 7) Located at the top of the page, click **restart required services**.

Result

Default Servers	
Registrar Host:	sip.registrarserver.com
Proxy Host:	sip.proxyserver.com
Messaging Server Host:	
Outbound Proxy Host:	

Next Step

[Configuring the Default Gateway](#) (p.9)

4.4 Configuring the Default Gateway

Before you start

Make sure your telephony interface has both Physical and Signalling up before starting.

Steps

- 1) Go to **SIP/Gateways**.
- 2) In the **Gateway Configuration** table, enter in the **Port** field, the listening port the Gateway will be using for SIP signaling if it is different from 5060.

Note: The default value is 0 which stands for 5060.

- 3) Click **Apply**.

Result

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

Next Step

[Authenticating the SIP Default Gateway \(p.10\)](#)

4.5 Authenticating the SIP Default Gateway

Steps

- 1) Go to **SIP/Authentication**.
- 2) In the **Authentication** table, click  located on the same line as the Default Gateway.
- 3) Set the following parameters:
 - a) Set **Validate Realm** to **Disable**.
 - b) Set the **User Name** to reflect your configuration.
 - c) Set the **Password** to reflect your configuration.
- 4) Click **Apply and Refresh Registration**.

Result

Authentication							
Priority	Criteria	Endpoint	Gateway	Username Criteria	Validate Realm	Realm	User Name Password
1	Gateway		SIP-default		Disable		\$Example \$Example

Next Step

[Restarting Services \(p.10\)](#)

4.6 Restarting Services

Steps

- 1) Go to **System/Services**.
- 2) In the **restart required services** table, click **restart required services**.

Next Step

[Creating a Route From BRI to Asterisk \(p.11\)](#)

5 Configuration of Routes

5.1 Creating a Route From BRI to Asterisk

Steps

- 1) Go to **Call Router/Route Config**.
- 2) In the **Routes** table
 - click  located on the same row as an existing route to add a route above or,
 - click  located at the bottom of the table to add a route at the end of the table.
- 3) In the **Create New Route** table, set the following parameters:
 - a) Set **Sources** to BRI.
 - b) Set **Criteria Property** to **None**.
 - c) Set **Destination** to **sip-default**.
- 4) Click **Save**.

Result

The route will be added to the **Routes** table.

Configure New Route		
	Value	Suggestion
Sources	isdn-Slot2/Bri0, isdn-Slot2/Bri1, isdn-Slot2/Bri2, isdn-Slot2/Bri3, isdn-Slot2/Bri4	--- Suggestion --- ▾
Criteria Property	None ▾	
Criteria Rule		--- Suggestion --- ▾
Transformations		--- Suggestion --- ▾
Signaling Properties		--- Suggestion --- ▾
Destination	sip-default	--- Suggestion --- ▾
Config Status		

Next Step

Creating a Hunt Group - BRI (p.12)

5.2 Creating a Hunt Group - BRI

Steps

- 1) Go to **Call Router/Route Config**.
- 2) In the **Hunt** table, click **+**
- 3) In the **Configure New Hunt** table, set the following:
 - a) Set the **Name** field to **BRI**.
 - b) Using the **Destinations'** drop box, add all the BRI ports used in your configuration.
 - c) Leave the other fields with their default value.
- 4) Click **Save**.

[illegible]

Result

Configure New Route		
	Value	Suggestion
Sources	sip-default	--- Suggestion ---
Criteria Property	None	
Criteria Rule		--- Suggestion ---
Transformations		--- Suggestion ---
Signaling Properties		--- Suggestion ---
Destination	hunt-BRI	--- Suggestion ---
Config Status		

Next Step

Registering a Unit to All Gateways (p.14)

5.4 Registering a Unit to All Gateways

Steps

- 1) Go to **SIP/Registrations**.
- 2) In the **Unit Registration** table, enter in the **User Name**, a username to uniquely identify the user in the domain.
- 3) From the **Gateway Name** selection list, choose **All**.
- 4) Repeat steps 2 (p.14) and 3 (p.14) for each SIP account/Registration needed.
- 5) Click **Apply and Refresh**.
- 6) Click **restart required services** located at the top of the page.

Result

The information will be displayed in the **Unit Registration Status** table.

Unit Registration			
Index	User Name	Gateway Name	
1	default_user	all	-
			+

Next Step

[Enabling Automatic Call Activation \(p.15\)](#)

5.5 Enabling Automatic Call Activation

Steps

- 1) Go to **Telephony/Services**.
- 2) In the **Select Endpoint** drop down menu, select the port you want to configure.
- 3) In the **Automatic Call** section, set the following parameters:
 - a) Set **Endpoint Specific** to **Yes**.
 - b) Set **Automatic Call Activation** to **Enable**.
 - c) Set the **Automatic Call Target** to the number of the IVR configured in the Asterisk server.
- 4) Repeat the steps **2** and **3** for the other ports you need to configure.

Result

Services Configuration	Unit Defaults	Endpoint Specific
General Configuration		
Endpoint Specific:		No
Hook Flash Processing:	Process Locally	Process Locally
Automatic Call		
Endpoint Specific:		Yes
Automatic Call Activation:	Disable	Enable
Automatic Call Target:		\$Example
Direct IP Address Call		
Direct IP Address Call Activation:	Disable	

Next Step

[Configuring DTMF Transport \(p.15\)](#)

5.6 Configuring DTMF Transport

Steps

- 1) Go to **Media/Misc**.

- 2) In the **DTMF Transport** table, set the **Transport Method** to the one set in the VoIP server.
 - a) **In-band**
 - b) **Out-of-Band using RTP** (RFC2833). This is usually the preferred method. Remember to set the payload type if different (e.g.: 96, 101 or 110 in some cases).
 - c) **Out-of-Band using SIP**. For Cisco or Avaya systems, from the **SIP Transport Method** field, use **Info DTMF Relay**. For legacy Nortel and others, leave **Draft Choudhuri SIP Info Digit 00**.
 - d) **Signaling Protocol Dependent**: choose this method if unsure. It will try to use the method negotiated by the VoIP server.
- 3) Click **Apply**.

Result

DTMF Transport			
Transport Method:	Out-of-Band using RTP	▼	
SIP Transport Method:	Draft Choudhuri SIP Info Digit 00	▼	
Payload Type:	101		

6 DGW Documentation

Mediatrix devices are supplied with an exhaustive set of documentation.

Mediatrix user documentation is available on the [Media5 Documentation Portal](#).

Several types of documents were created to clearly present the information you are looking for. Our documentation includes:

- **Release notes:** Generated at each GA release, this document includes the known and solved issues of the software. It also outlines the changes and the new features the release includes.
- **Configuration notes:** These documents are created to facilitate the configuration of a specific use case. They address a configuration aspect we consider that most users will need to perform. However, in some cases, a configuration note is created after receiving a question from a customer. They provide standard step-by-step procedures detailing the values of the parameters to use. They provide a means of validation and present some conceptual information. The configuration notes are specifically created to guide the user through an aspect of the configuration.
- **Technical bulletins:** These documents are created to facilitate the configuration of a specific technical action, such as performing a firmware upgrade.
- **Hardware installation guide:** They provide the detailed procedure on how to safely and adequately install the unit. It provides information on card installation, cable connections, and how to access for the first time the Management interface.
- **User guide:** The user guide explains how to customise to your needs the configuration of the unit. Although this document is task oriented, it provides conceptual information to help the user understand the purpose and impact of each task. The User Guide will provide information such as where and how TR-069 can be configured in the Management Interface, how to set firewalls, or how to use the CLI to configure parameters that are not available in the Management Interface.
- **Reference guide:** This exhaustive document has been created for advanced users. It includes a description of all the parameters used by all the services of the Mediatrix units. You will find, for example, scripts to configure a specific parameter, notification messages sent by a service, or an action description used to create Rulesets. This document includes reference information such as a dictionary, and it does not include any step-by-step procedures.

7 Copyright Notice

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