Basic BRI Mediatrix Unit Configuration with Asterisk

Mediatrix Unit with BRI Card

Revision 02

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Table of Contents

Mediatrix unit with Asterisk

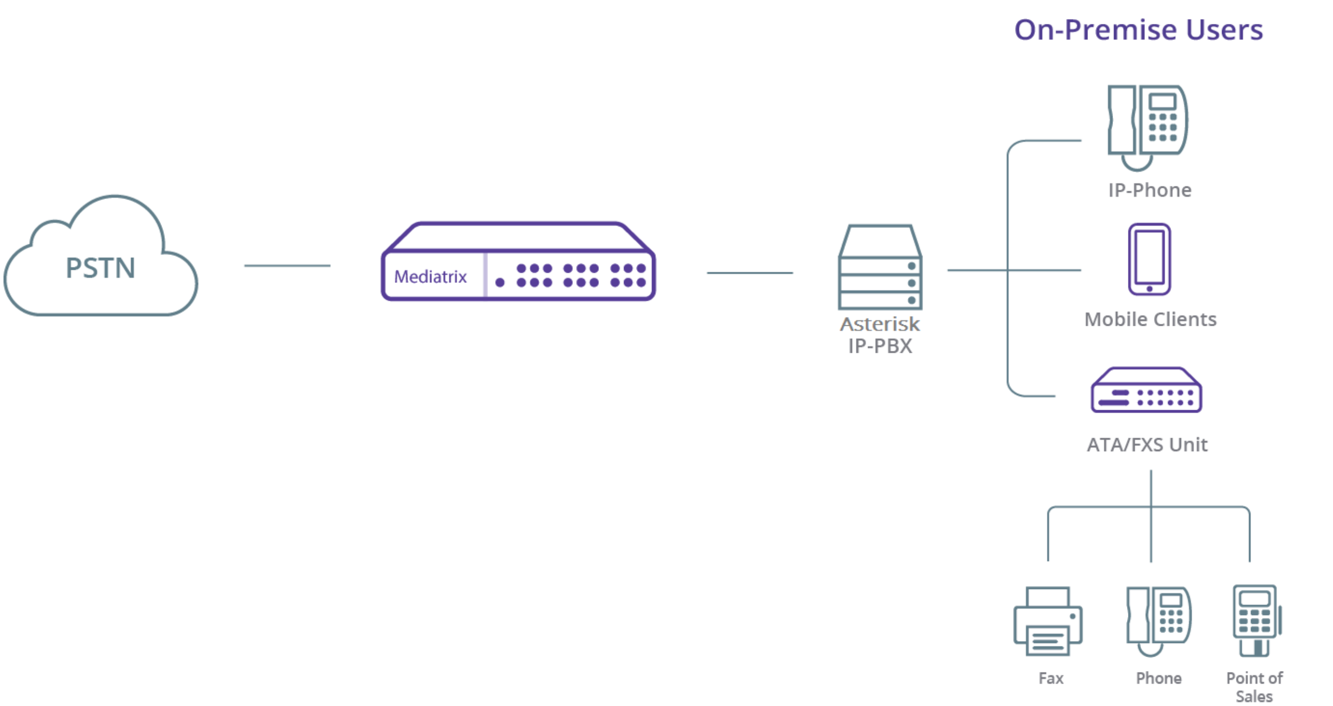
This document outlines the configuration steps required to connect a Mediatrix unit to an Asterisk open-source telephone system.

In this scenario, the Mediatrix 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



Requirements

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

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.

Note: These configuration notes can be successfully executed, provided your unit is using the Factory default settings.

Information	Value	Used in Step
IP address of unit		Configuring the Asterisk - PBX Trunk (p. 7) and Configuring the Asterisk - PSTN Lines (p. 8)
Listening port of unit		Configuring the Asterisk - PBX Trunk (p. 7) and Configuring the Asterisk - PSTN Lines (p. 8)
The SIP username used for calls coming from the PSTN		Configuring the Asterisk - PSTN Lines (p. 8)
The SIP username used for calls coming from the PBX		Configuring the Asterisk - PSTN Lines (p. 8)
SIP Username and Password to authenticate the gateway		Authenticating the SIP Default Gateway (p. 12)
Asterisk server IP address		
Asterisk server SIP listening port		
DTMF transport method		Configuring the Asterisk - PBX Trunk (p. 7) and Configuring the Asterisk - PSTN Lines (p. 8) and Configuring DTMF Transport (p. 19)

Configuration of the PBX Trunk

Configuring the Asterisk - PBX Trunk

Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

Steps

1) In the **sip.config** 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 Mediatrix unit
   d) port= listening port of the Mediatrix 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. 9) 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 Standard Fax Configuration document.

Next Step

Configuring the Asterisk - PSTN Lines (p. 8)

Configuring the Asterisk - PSTN Lines

Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

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. 9) 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 Servers (p. 10)

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 Mediatrix unit	This means the extension will not register to the Asterisk server. This is the IP address of the Mediatrix unit.
port=xxx	Port used for requests to and from the Mediatrix unit.
nat=no	The Mediatrix unit is not behind a NAT.
qualify=no	No keep alive is used.
canreinvite=no	No Re-Invite 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 Mediatrix Unit.
secret=TrunkPassword	The SIP authentication password

Extension	Description
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 Mediatrix unit .

Configuring the Default Servers

Before you start

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

Information

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

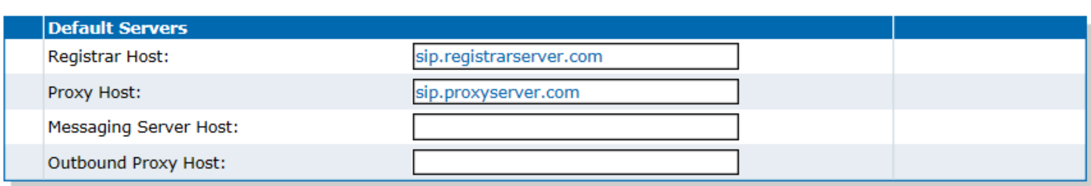
Steps

1) Go to **SIP/Servers**.
2) In the **Registrar Host** field, indicate the server IP address or FQDN to use for this gateway.
3) In the **Proxy Host** field, indicate the server IP address or FQDN to use for this gateway.
4) In the **Messaging Server Host** field, indicate the server IP address or FQDN to use for this gateway, 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



Next Step

Configuring the Default Gateway (p. 11)

Configuring the Default Gateway

Before you start

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

Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

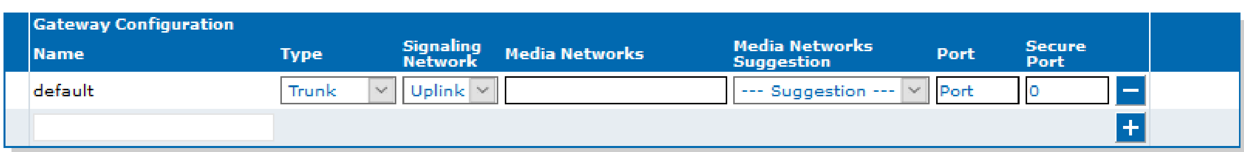
Steps

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



Next Step

Authenticating the SIP Default Gateway (p. 12)

Authenticating the SIP Default Gateway

Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

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. 13)

Restarting Services

Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

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. 14)

Configuration of Routes

Creating a Route From BRI to Asterisk

Information

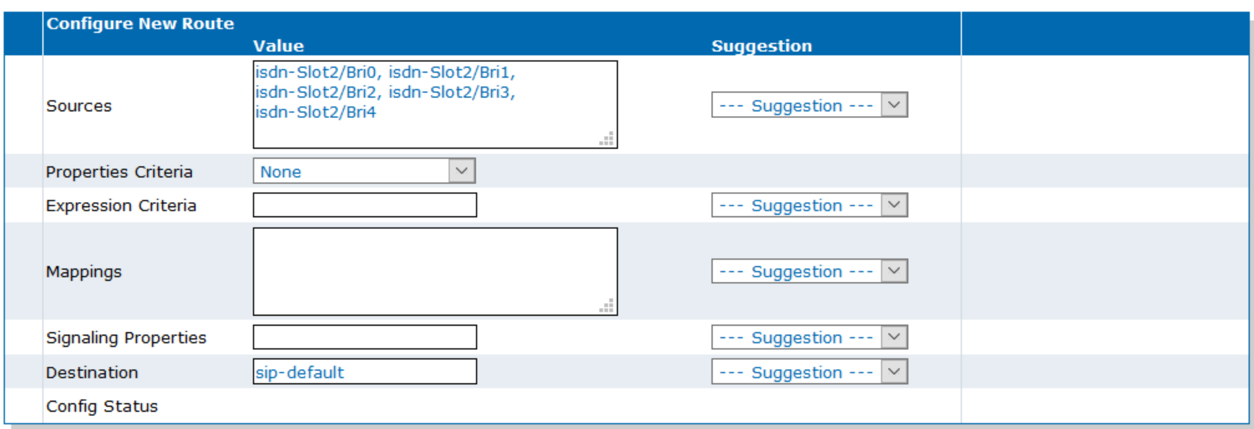
If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

Steps

1) Go to **Call Router**/**Route Config**.
2) In the **Route** 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 **Configure Route End** table, set the following parameters:
   a) Set **Sources** to BRI.
   b) Set **Properties Criteria** to **None**.
   c) Set **Destination** to **sip-default**.
4) Click **Save**.

Result

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



Next Step

Creating a Hunt Group - BRI (p. 15)

Creating a Hunt Group - BRI

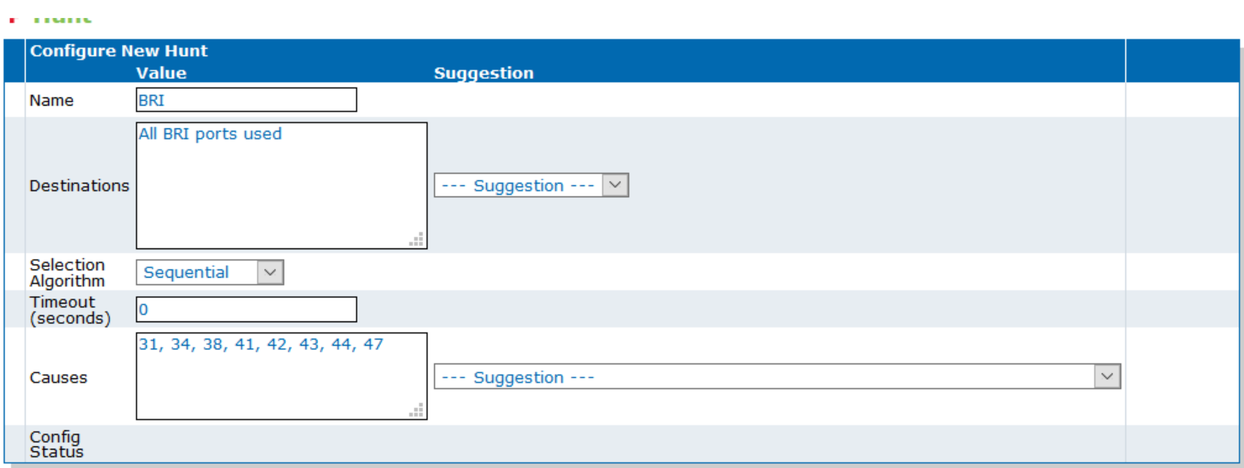
Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

Steps

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

Result



Next Step

Creating a Route from Asterisk to the BRI Hunt Group (p. 16)

Creating a Route from Asterisk to the BRI Hunt Group

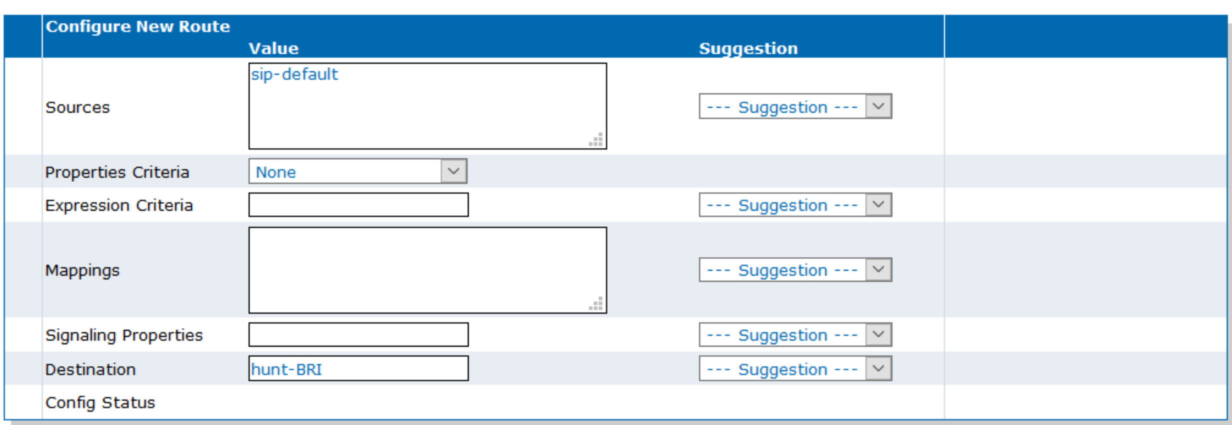
Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

Steps

1) Go to **Call Router**/**Route Config**.
2) In the **Route** 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 **Configure Route End** table, set the following parameters:
   a) Set **Sources** to **sip-default**.
   b) Set **Properties Criteria** to **None**.
   c) Set **Destination** to **hunt_BRI**.
4) Click **Save**.

Result



Next Step

Registering a Unit to All Gateways (p. 17)

Registering a Unit to All Gateways

Before you start

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

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 the Sip gateway the user will be assigned to.
4) Click **Apply and Refresh**.
5) 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
2	other_user	all

Next Step

Enabling Automatic Call Activation (p. 18)

Enabling Automatic Call Activation

Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

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. 19)

Configuring DTMF Transport

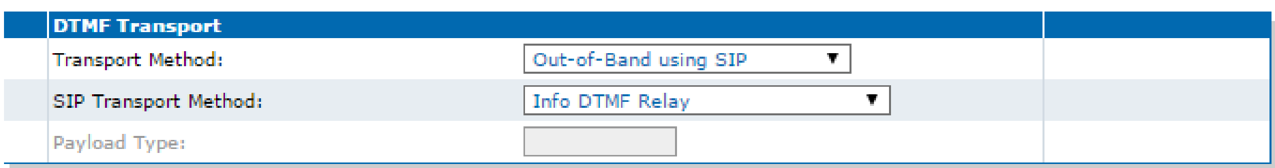
Information

If you are not familiar with the meaning of the fields, click **Show Help**, located at the upper right corner of the Web page, to display field description when hovering over the field name.

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) Inband**Inband**
   b) **Out-of-Band using RTP** (RFC2833)
   c) **Out-of-Band using SIP**, if selected, set **SIP Transport Method** to **Info DTMF Relay**
3) Click **Apply**.

Result



Documentation

Mediatrix units are supplied with an exhaustive set of documentation.

Mediatrix user documentation is available at Mediatrix Documentation.

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 customize 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 is an exhaustive document 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.

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