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Enabling Live Communications at the Edge of IP Networks

## Configuration Notes 279

## Configuring the Mediatrix Analog Units to work with the Asterisk Telephone System

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## Introduction

This document outlines the configuration steps to set up Mediatrix analog units with the Asterisk open-source telephone system. It assumes that you have an Asterisk server properly installed with the necessary modules.

## Features Supported

The following features are supported by both Asterisk and Mediatrix analog units:

- RFC 2833, SIP INFO, and inband DTMF transports
- SIP Authentication
- Blind and Supervised Call Transfer
- Call Forward On Busy/On No Answer/Unconditional
- Call Waiting (via Asterisk only)
- Voice Mail

- Conference
- Music on Hold
- Caller ID
- PSTN Trunk Calls (via Mediatrix 1204)
- G.711 (recommended), G.723.1 and G.726 codecs
- Fax transmission
- IVR

## Asterisk Configuration

The following sections describe special configuration you must perform in Asterisk in order to properly work with the Mediatrix analog units. The configuration parameters are located in various configuration files.

### Creating an Extension

In Asterisk, an extension is the equivalent of the SIP user in the Mediatrix units.

1. In the */etc/asterisk/sip.conf* file, scroll to the bottom of the file and add the following:

```
[101]
type=friend
host=dynamic
mailbox=101@VoiceServer
nat=yes
qualify=yes
canreinvite=no
dtmfmode=info
context=sip
secret=num101
```

| Parameter | Description |
|---|---|
| [101] | This is the extension number. Use a unique number. The Mediatrix unit will use this number to authenticate to the system and users will dial it to ring the Mediatrix unit. |
| type | Attribute of the SIP object. There are three values available:<br><br>**peer**: A SIP entity to which Asterisk sends calls (a SIP provider for example). If you want a user (extension) to have multiple phones, define an extension that calls two SIP peers. The peer authenticates at registration.<br><br>**user**: A SIP entity which places calls through Asterisk.<br><br>**friend**: An entity which is both a user and a peer. This make sense for most desk handsets and other devices.<br><br>If a peer is defined with **host=dynamic** it is allowed to register with Asterisk to tell Asterisk where it can be found (IP address/host name) and that it is reachable from now on. |
| host | How to find the client - IP # or host name. Using the keyword **dynamic** indicates that the phone registers itself. |
| mailbox | Refers to the corresponding mailbox of the context in the *voicemail.conf* file. |

| Parameter | Description |
|---|---|
| nat | When entering **yes**, Asterisk will change the behaviour, adressing etc, of communication with the client (SIP UA) that is behind a NAT device, to make communication possible. |
| qualify | Entering **yes** indicates that Asterisk keeps a UDP session open to a device that is located behind a network address translator (NAT). This can be used in conjunction with the **nat=yes** setting. |
| canreinvite | Entering **no** indicates that Asterisk does not issue a reinvite to the client and acts as the RTP portal. |
| dtmfmode | You have the choice between *rfc2833*, *info*, and *inband*. Mediatrix recommends to use the **info** mode. You will have to enter similar information in the Mediatrix unit. See SIP Configuration on page 9 for more details. |
| context | Refers to the context to which the extension belongs as defined in the *extensions.conf* file (see Step 2 for details). |
| secret | If you want to use authentication for this extension, enter the proper information in the *secret* field. You will have to enter similar information in the Mediatrix unit. See SIP Configuration on page 9 for more details. |

2. In the */etc/asterisk/extensions.conf* file, add the following:

```
[sip]
exten=>101,1,Dial(SIP/101,20,RtT)
exten=>101,2,Voicemail,u101
exten=>101,102,Voicemail,b101
exten=>981,1,VoicemailMain
```

Notes:

The **T** and **t** characters are used to allow blind and supervised call transfer. When the *T* or *t* option is in the Dial command, Asterisk will ignore the *canreinvite=yes* command. It will thus be used as a RTP portal. See Blind Transfer and Supervised (Attended) Transfer for more details.

You can also replace the last **101** extension of the expression with the *${EXTEN}* variable (for instance, *exten=>101,1,Dial(SIP/${EXTEN},20,RtT)*.

3. In the */etc/asterisk/voicemail.conf* file, add the following:

```
[VoiceServer]
101=>1234,John Smith,jsmith@test.com
```

4. Execute the following command on the server to reload the configuration:

```
asterisk -rx "reload"
```

## Authentication via Password

The following are the tasks you must perform in Asterisk if you want to have authentication via password:

1.  In the **[sip]** section of the */etc/asterisk/extensions.conf* file, modify the section as follows (in red bold):

    ```
    [sip]
    exten=>101,1,Authenticate(1234)
    exten=>101,2,Dial)SIP/101,20,RtT)
    exten=>101,3,Voicemail,u101
    ;exten=>101,102,Voicemail,b101
    exten=>981,1,VoicemailMain
    ```

    Notes:

    **1234** is the password to enter.

    Be sure to put the **exten=>101,102,Voicemail,b101** line in comment by adding a "**;**" at the beginning of the line.

2.  Execute the following command on the server to reload the configuration:

    ```
    asterisk -rx "reload"
    ```

**Note**: You can also perform authentication via voice mail. See the Asterisk documentation for more details.

## Blind Transfer

Asterisk has a built-in blind transfer feature. To activate this feature, do the following:

1.  In the **[featuremap]** section of the */etc/asterisk/features.conf* file, remove the "**,**" at the beginning of the following line:

    ```
    blindxfer=>#1
    ```

    Note that **#1** corresponds to the default digits a user must dial to transfer a call. You can put anything in place of "#1".

    During a call, the calling user simply has to dial **#1** followed by the proper telephone number to perform a blind call transfer.

2.  Execute the following command on the server to reload the configuration:

    ```
    asterisk -rx "reload"
    ```

## Supervised (Attended) Transfer

Asterisk has a built-in supervised transfer feature. The following are the tasks you must perform in Asterisk:

1.  In the **[featuremap]** section of the */etc/asterisk/features.conf* file, remove the "**,**" at the beginning of the following line:

    ```
    atxfer => *2
    ```

    Note that **\*2** corresponds to the default digit a user must dial to signal a call transfer. For instance, with an active call, the user must dial **\*2** followed by the proper telephone number to perform a supervised call transfer. You can put anything in place of "*2".

2.  Execute the following command on the server to reload the configuration:

    ```
    asterisk -rx "reload"
    ```

## Conference

If you want to add a password to the conference, proceed as follows:

**1.** In the **[rooms]** section of the */etc/asterisk/meetme.conf* file, add the following line:

```
conf => 8101,1234
```

where **1234** is the password you want to use for the conference on extension *101*. The *101* extension must exist.

**2.** In the */etc/asterisk/extensions.conf* file, add the following section:

```
[conference]
exten=>8101,1,Meetme(8101)
```

Notes:

The first *8101* corresponds to the number to dial.

The second *8101* corresponds to the room number.

**3.** In the **[sip]** section of the */etc/asterisk/extensions.conf* file, add the following line:

```
include => conference
```

**4.** Execute the following command on the server to reload the configuration:

```
asterisk -rx "reload"
```

**Note**: If the conference number is invalid, make sure that the *ztdummy* module is properly installed and configured.

## Outgoing Calls via a Mediatrix 1204

You must create a new SIP trunk:

**1.** In the */etc/asterisk/sip.conf* file, scroll to the bottom of the file and add the following:

```
[1204Gateway]
type=peer
host=192.168.0.10
mask=255.255.255.255
canreinvite=yes
dtmfmode=info
context=sip
```

| Parameter | Description |
|---|---|
| [1204Gateway] | Name of the trunk. |
| type | Attribute to SIP object. There are three values available:<br><br>**peer**: A SIP entity to which Asterisk sends calls (a SIP provider for example). If you want a user (extension) to have multiple phones, define an extension that calls two SIP peers. The peer authenticates at registration.<br><br>**user**: A SIP entity which places calls through Asterisk.<br><br>**friend**: An entity which is both a user and a peer. This make sense for most desk handsets and other devices. |
| host | Enter the address of the Mediatrix 1204. |
| mask | Enter the network mask of the Mediatrix 1204. |
| canreinvite | Entering **yes** indicates that once the call has been accepted, Asterisk sends another (re)INVITE message to the clients with the information necessary to have the two clients send the media streams directly to each other. |
| dtmfmode | You have the choice between *rfc2833*, *info*, and *inband*. Mediatrix recommends to use the **info** mode. You will have to enter similar information in the Mediatrix unit. See SIP |

| Parameter | Description |
|-----------|-------------|
|  | Configuration on page 9 for more details. |
| context | Refers to the context to which the extension belongs as defined in the *extensions.conf* file (see Creating an Extension  for details). |

2.  In the **[sip]** section of the */etc/asterisk/extensions.conf* file, add the following calling route:

```
exten=>_1418XXXXXXX,1,Dial(SIP/${exten:4}@1204Gateway)
```

Notes:

**1** represents the priority of the route. With several routes, define the priority carefully. For routes that match long distance numbers, for example, you'd want to pick the cheapest routes for long distance (ie, VoIP trunks first) followed by more expensive routes (POTS lines).

**exten:4** indicates that when dialing, Asterisk will strip the first four digits of the number.

Add any other calling rule as required by your system.

3.  Execute the following command on the server to reload the configuration:

```
asterisk -rx "reload"
```
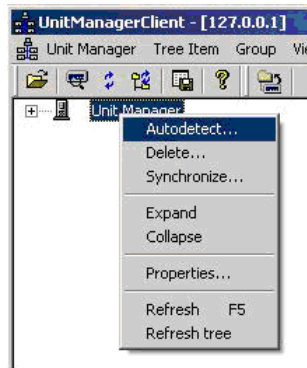
## Mediatrix Units Configuration

The following sections describe special configuration you must perform in the Mediatrix units in order to properly work with Asterisk.

### Running the Unit Manager Network Software

The Unit Manager Network (UMN) software is a configuration and management tool for Mediatrix units. The UMN is provided on the CD included with the unit. It has a default 3-units limit upon installation. This will suffice for most configurations.
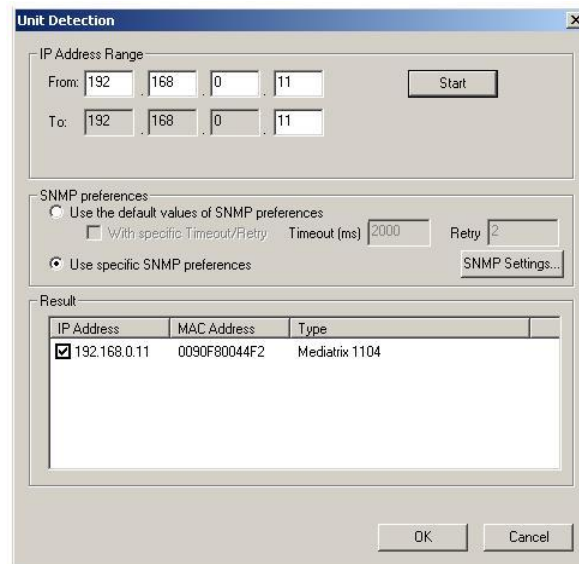
It is recommended that the administrator uses the UMN to configure the Mediatrix units. Please refer to the UMN Quick Start guide for the installation of the software. Once the UMN software has been installed on your PC, proceed with the following steps.

1.  Start the UMN by selecting  Start Menu > Programs > Unit Manager Network 3.2 > Unit Manager Network.

2.  In the *Administrator* login window (Connect to Unit Manager), a User Name and Password are not required. Click **OK** to proceed.

3.  On the left pane, right-click the *Unit Manager* level, then select **Autodetect**.



4.  Set the *IP Address Range* to minimize the time it takes to auto-detect the unit. Click **Start** to begin Mediatrix unit detection. When the unit is detected, the *Result* section lists the unit.
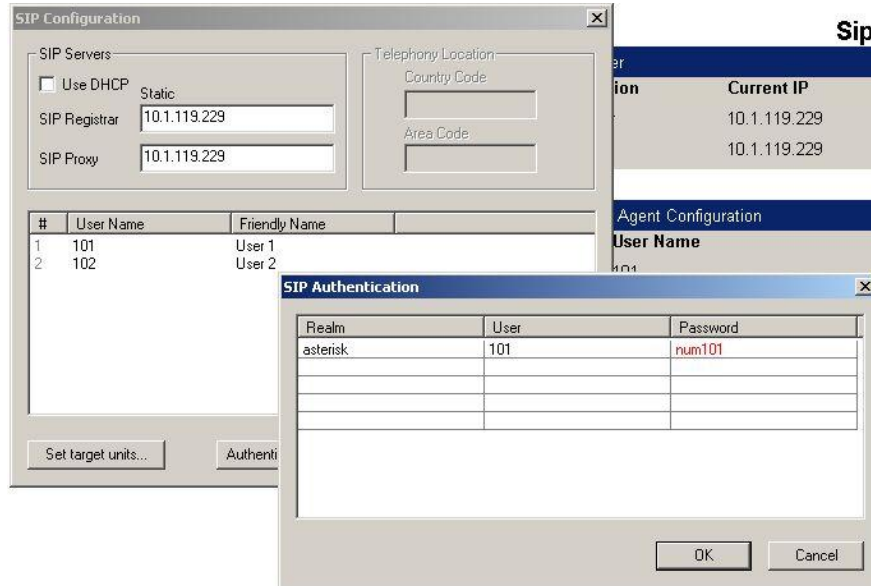
5. Select the unit and click **OK**. If no DHCP server is used in your subnet, you must connect one unit at a time since they will start by using the default IP address 192.168.0.1 after a recovery reset. You will have to set a different static IP address for every unit. Please see the **1100 series and 1204 SIP Quick Start Guide** for more details on initial setup.
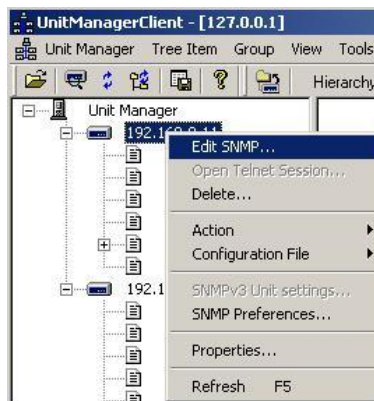
## SIP Configuration

In Asterisk, an extension is the equivalent of the SIP user in the Mediatrix units. You must match the extension you have created in Asterisk in section Creating an Extension on page 3.

1. Once the Mediatrix unit has been auto-detected in the UMN, it should appear under the *Unit Manager* level. Right-click the *Sip* level of the Mediatrix unit and select the **Edit** menu.

2. In the *SIP Registrar* and *SIP proxy* fields, enter the address of the PC that hosts Asterisk.

3. In the *User Name* column, enter a user name as defined in Asterisk.

4. If you have entered a secret in the Asterisk authentication information, click the **Authentication** button.

5. Enter **asterisk** as the *Realm*.

6. In the *Password* column, enter the same secret as set in Asterisk.
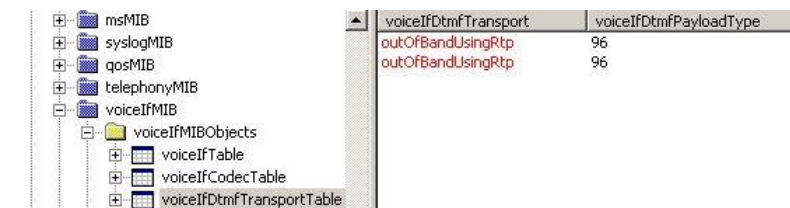
7. Click **OK**.

8. Click **OK** once more.

**9.** Right-click the unit and select **Edit SNMP**.



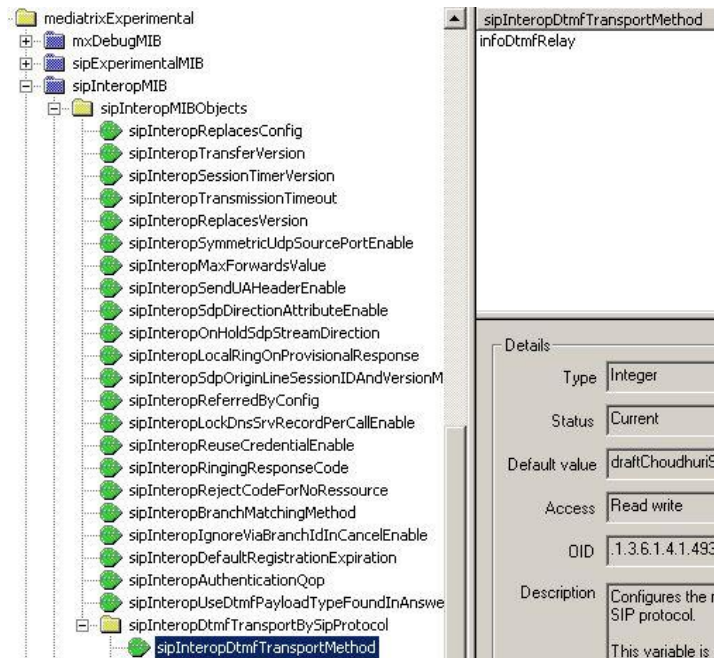**10.** On the top menu bar, check the option **Automatic GET**.

**11.** Go to *iso>org>dod>internet>private>enterprises>mediatrix>mediatrixConfig>voiceIfMIB>voiceIfMIBObjects> voiceIfDtmfTransportTable>***voiceIfDtmfTransport** and set this parameter according to the DTMF transport mode you have defined in the AMP.

- o   If you have used *rfc2833*: set the parameter to **outOfBandUsingRtp**.

- o   If you have used *info*, set the parameter to **outOfBandUsingSignalingProtocol**.

- o   If you have used *inband*, set the parameter to **inBand**.



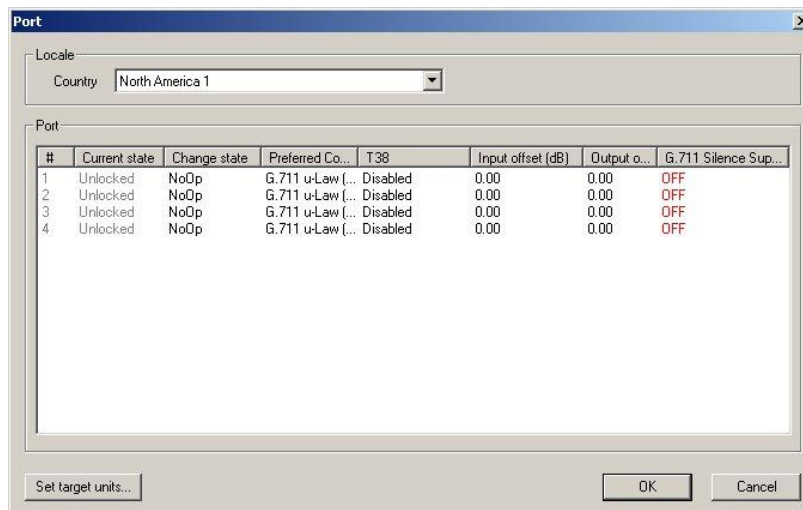**Note** : Mediatrix recommends to use the **outOfBandUsingSignalingProtocol** method.

**12.** If you have used the *info* DTMF transport, go to *iso>org>dod>internet>private>enterprises>mediatrix>
mediatrixExperimental>sipExperimentalMIB>sipExperimentalMIBObjects>sipInteropDtmfTransportBySipProtocol>*
**sipInteropDtmfTransportmethod** and set this parameter to **infoDtmfRelay**.



## Silence Suppression

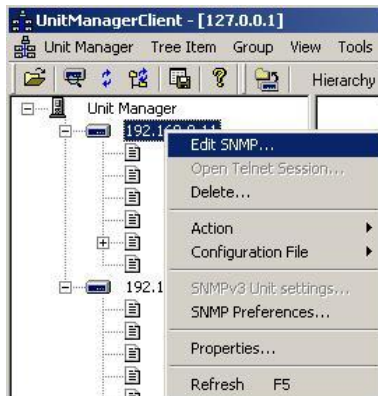You must turn off the Silence Suppression feature in the Mediatrix units.

**1.** In the UMN, right-click the *Port* level of the Mediatrix unit and select the **Edit** menu.

**2.** In the *G.711 Silence Suppression* column, change all settings to **OFF**.

**3.** Click **OK**.

## Conference

You must set the ptime to 20 ms.

1.  In the UMN, right-click the unit and select **Edit SNMP**.



2.  Go to *iso>org>dod>internet>private>enterprises>mediatrix>mediatrixConfig>voiceIfMIB>voiceIfMIBObjects> voiceIfCodectTable>***voiceIfCodecPcmuMinPTime** and set this parameter to **20**.

3.  Set the  voiceIfCodecPcmuMaxPTime parameter to **20**.



# References

For more information on the Asterisk configuration parameters and the Mediatrix products, visit these links :

http://www.voip-info.org/

http://www.mediatrix.com/products.php