



Enabling Live Communications at the Edge of IP Networks

Configuration Notes 290

Configuring Mediatrix 41xx FXS Gateway with the Asterisk IP PBX System

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Proprietary

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Introduction

This document outlines the configuration steps to set up Mediatrix 41xx series FXS gateways with the Asterisk open-source telephone system. It assumes that you have an Asterisk server properly installed with the necessary modules. If you need technical assistance to configure your Asterisk server, the Mediatrix technical team can provide appropriate support to help you realizing your VoIP projects.

About Mediatrix 41xx Series FXS Gateways

The Mediatrix® 41xx Series products are high-quality and cost-efficient VoIP gateways connecting larger branch offices and multi-tenant buildings to an IP network, while preserving investment in analog telephones and faxes.

The Mediatrix® 41xx access devices allow Service Providers to deploy rapidly and economically their solutions in medium-size premises and they are the ideal solution for branch office connectivity to larger private networks. The following Mediatrix 41xx models are available:

- **4102** – Residential/Enterprise 2-port FXS gateway
- **4104** – Enterprise 4-port FXS gateway
- **4108** – Enterprise 8-port FXS gateway
- **4116** – Enterprise 16-port FXS gateway
- **4124** – Enterprise 24-port FXS gateway

Benefits of using the Mediatrix 41xx over other Asterisk FXS solutions:

- Superior Voice quality with dedicated DSP hardware
- Adaptive Jitter Buffer, G.168 Echo Cancellation
- T.38 Fax Relay support
- Can be used to extend the Asterisk system in remote locations
- Highly flexible, easily adaptable to VoIP systems other than Asterisk
- Simple configuration to work with Asterisk



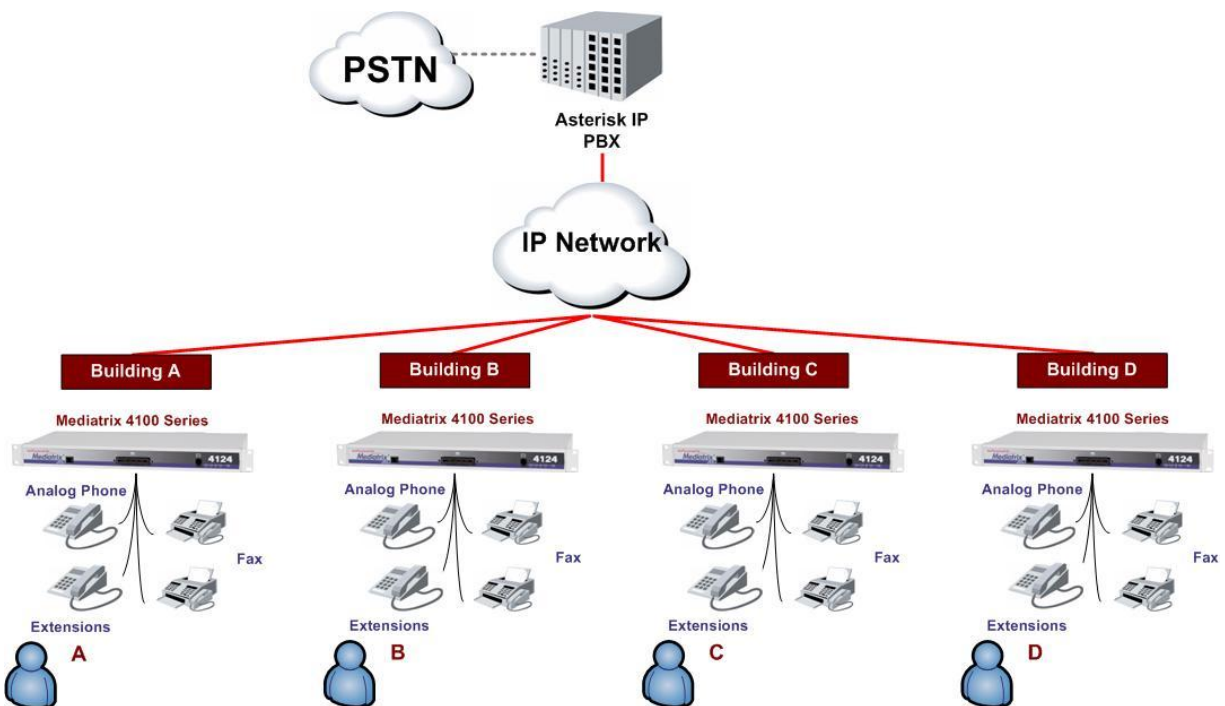
Application Scenario

This is a typical deployment scenario of Mediatrix 41xx units in a SIP-based MTU/MDU environment. The units provide connectivity to analog phones and fax machines and analog trunking to legacy PBX/KSU (for small business) in each building floor, whereas the Asterisk IP PBX provides call control and telephony services for all gateways.

FXS Gateway (41xx) – connectivity to analog phones and fax machines, analog trunking to legacy PBX/KSU.

The Asterisk IP PBX – provides:

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



Features Supported

The following features are supported by both Asterisk and Mediatrix 41xx:

- 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 Call
- Music on Hold
- Caller ID
- G.711 (recommended), G.723.1 and G.726 codecs
- Fax transmission
- IVR

Versions Supported

This configuration note was written and validated using the following platforms and versions.

- Asterisk: 1.2.4 running on RedHat Linux 9
- Mediatix 41xx: SIP 5.0.18.113

The following configuration notes are not a substitute for the Mediatix Administration Documentation for Mediatix 41xx products. Please have the following manuals available for reference:

- Mediatix 41xx SIP Reference Manual
- Mediatix 41xx SIP Quick Start
- Mediatix Unit Manager Network Administration Manual
- All relevant Asterisk Manuals

Asterisk Configuration

The following sections describe special configuration you must perform in Asterisk in order to properly work with the Mediatix 41xx. 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 Mediatix 41xx.

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

```
[101]
type=friend
host=dynamic
nat=yes
qualify=yes
canreinvite=no
dtmfmode=rfc2833
context=sip
username=101
secret=num101
```

| Parameter | Description |
|-------------|---|
| [101] | This is the extension number. Use a unique number. The Mediatix 41xx will use this number to authenticate to the system and users will dial it to ring the Mediatix 41xx. |
| type | Attribute of the SIP object. There are three values available: <ul style="list-style-type: none"> ▪ 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. ▪ user: A SIP entity that places calls through Asterisk. ▪ friend: An entity that is both a user and a peer. This makes sense for most desk handsets and other devices. <p>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.</p> |
| host | How to find the client - IP # or host name. Using the keyword dynamic indicates that the phone registers itself. |
| nat | When entering yes , Asterisk will change the behaviour, addressing, 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. The Mediatix 41xx supports both yes and no options. |

| Parameter | Description |
|-----------|--|
| dtmfmode | You have the choice between <i>rfc2833</i> , <i>info</i> , and <i>inband</i> . You will have to enter similar information in the Mediatrix 41xx. |
| context | Refers to the context to which the extension belongs as defined in the <i>extensions.conf</i> file (see Step 2 for details). |
| secret | If you want to use authentication for this extension, enter the proper information in the <i>secret</i> field. You will have to enter similar information in the Mediatrix 41xx. See SIP AuthenticationError! Reference source not found. on page 10 for more details. |

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

```
[sip]
exten=>101,1,Dial(SIP/101)
```

- Execute the following command on the server to reload the configuration:

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

Mediatrix 41xx Configuration

The following sections describe special configuration you must perform in the Mediatrix 41xx in order to properly work with Asterisk.

All models in the Mediatrix 41xx series feature an embedded Web server. Most of the commonly used parameters are accessible from the web interface. The Mediatrix Unit Manager Network (UMN) software is needed if access to full unit configuration is required. The UMN can be downloaded from the Mediatrix Portal:

<https://support.mediatrix.com/DownloadPlus/Download.asp>

It has a default 3-units limit upon installation. This will suffice for most configurations. Additional unit license can be purchased. Please contact your Mediatrix reseller for more details.

Using the Web interface

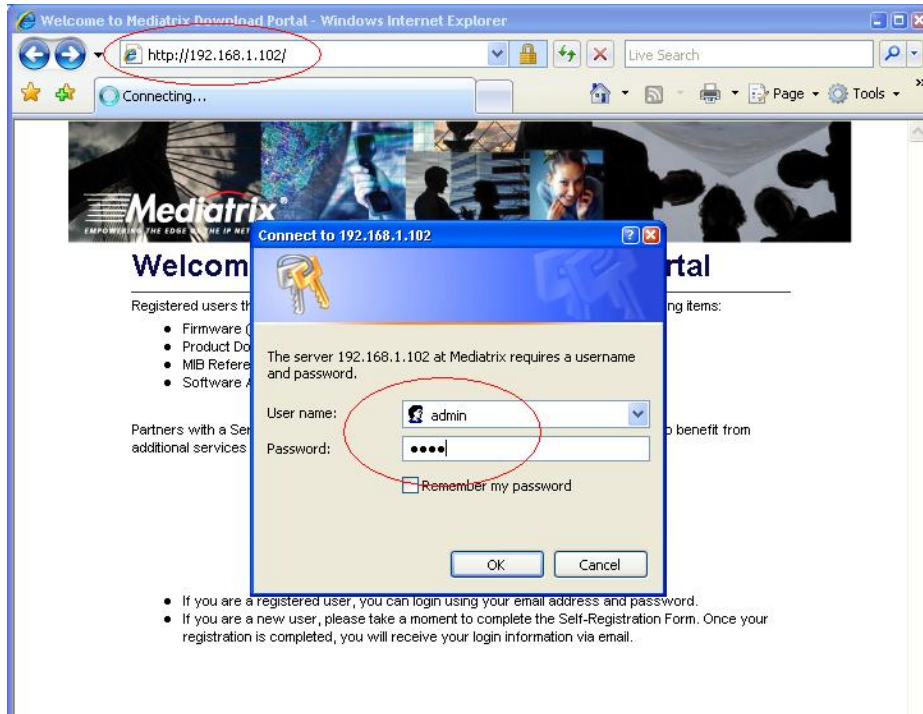
The web interface may be used to:

- View the status of the Mediatrix 41xx.
- Set numerous parameters of the Mediatrix 41xx.

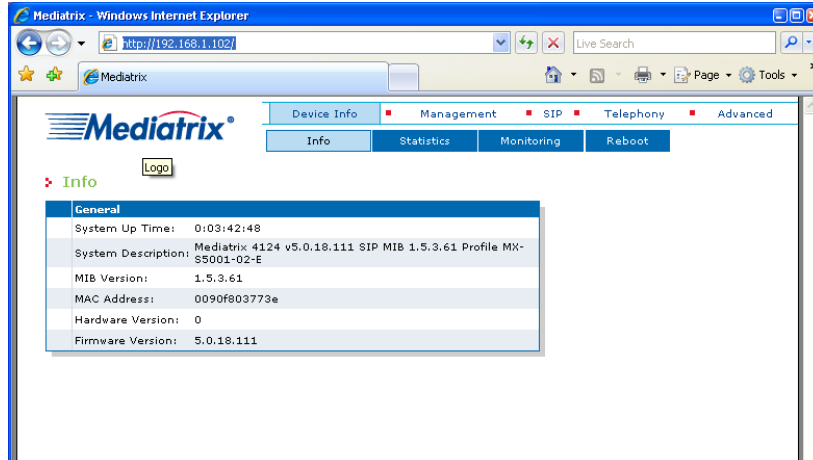
Mediatrix recommends that you use the latest version of the Microsoft® Internet Explorer web browser to properly access the web interface. To use the web interface configuration:

- In the web browser's address field, type the IP address of the Mediatrix 41xx (if you have performed a recovery mode, this is 192.168.0.1). The unit's IP address can be found by dialing ****0** from a phone connected to port 2 or above.

2. Enter the default login name **admin** and password **1234**.



3. The system information screen then appears, giving information about the firmware version, hardware revision, system up-time and MAC address.



Network Parameters Configuration

Network parameters are located in the *Management. / Network Settings* page.

1. Click **Management** and **Network Settings** to go the *Network Settings* web page.

The screenshot displays the Mediatrix web interface for Network Settings. The navigation menu at the top includes 'Device Info', 'Management', 'SIP', 'Telephony', and 'Advanced'. The 'Management' menu is expanded to show 'Network Settings', 'Configuration File', 'Firmware Download', and 'Reboot'. The 'Network Settings' section is active and contains several sub-sections:

- Current Configuration:** Shows IP Address Source (DHCP), IP Address (192.168.1.102), Subnet Mask (255.255.255.0), Default Router (192.168.1.1), Primary DNS (192.168.1.1), Secondary DNS (0.0.0.0), and DHCP Server IP Address (192.168.1.1).
- Ethernet:** Network Port Speed is set to 'autoDetect'.
- Network Settings:** IP Address Source is set to 'Static' (indicated by a red arrow and text: 'Click here if static IP is used'). Fields for IP Address (192.168.0.1), Subnet Mask (255.255.255.0), Default Router (192.168.0.10), Primary DNS (192.168.0.10), Secondary DNS (192.168.0.10), and SNMP Port (161) are visible. A green arrow points to these fields with the text: 'Static IP, subnet mask, default gateway, DNS'.
- SNTP Info:** SNTP Source is DHCP, SNTP Port is 123, and SNTP Host is 0.0.0.0.
- SNTP:** SNTP Enable is set to 'Enable'. SNTP Source is DHCP. SNTP Host is 192.168.0.10, SNTP Port is 123, SNTP Timezone is EST/SD/ST4.M4.1.0/02.0, SNTP Synchronization Period is 1440, and SNTP Synchronization Period on Error is 60. A green arrow points to the SNTP Host field with the text: 'SNTP Server here'. The 'Submit' button at the bottom is circled in red.

2. In the *IP Address Source* option, select **DHCP** or **Static**.
DHCP is the default selection.
3. If you are using a Static IP address, select **Static** and enter the IP address, subnet mask, default router and DNS server IP in the proper fields.
4. If SNTP is required, set the *SNTP Enable* option to **Enable** and enter the appropriate SNTP server IP address in the *SNTP Host* field.
5. Click **Submit** to apply the changes.

SIP Configuration

In Asterisk, an extension is the equivalent of the SIP user in the Mediatrix 41xx. You must match the extension you have created in Asterisk in section [Creating an Extension](#) on page 5.

1. Click **SIP** and **Configuration** to go the *SIP Configuration* web page.

SIP Info

| | |
|----------------------|---------|
| SIP Server Source: | DHCP |
| Registrar Host: | 0.0.0.0 |
| Registrar Port: | 0 |
| Proxy Host: | 0.0.0.0 |
| Proxy Port: | 0 |
| Outbound Proxy Host: | 0.0.0.0 |
| Outbound Proxy Port: | 0 |

SIP Configuration

SIP Server Source: Static ← Click here if SIP Setting is used DHCP

SIP Port:

Registrar Host: ← SIP Registrar IP and SIP Proxy IP (IP of the Asterisk server)

Registrar Port:

Proxy Host:

Proxy Port:

Outbound Proxy Host:

Outbound Proxy Port:

Unregistered Port Behavior:

SIP User Configuration

| Port | User Name | Friendly Name | Other Accepted Username |
|------|--------------------------------------|------------------------------------|-------------------------|
| 1 | <input type="text" value="101"/> | <input type="text" value="John"/> | <input type="text"/> |
| 2 | <input type="text" value="102"/> | <input type="text" value="David"/> | <input type="text"/> |
| 3 | <input type="text" value="3330003"/> | <input type="text"/> | <input type="text"/> |
| 4 | <input type="text" value="3330004"/> | <input type="text"/> | <input type="text"/> |
| 5 | <input type="text" value="3330005"/> | <input type="text"/> | <input type="text"/> |
| 6 | <input type="text" value="3330006"/> | <input type="text"/> | <input type="text"/> |
| 7 | <input type="text" value="3330007"/> | <input type="text"/> | <input type="text"/> |
| 8 | <input type="text" value="3330008"/> | <input type="text"/> | <input type="text"/> |
| 9 | <input type="text" value="3330009"/> | <input type="text"/> | <input type="text"/> |
| 10 | <input type="text" value="3330010"/> | <input type="text"/> | <input type="text"/> |
| 11 | <input type="text" value="3330011"/> | <input type="text"/> | <input type="text"/> |
| 12 | <input type="text" value="3330012"/> | <input type="text"/> | <input type="text"/> |
| 13 | <input type="text" value="3330013"/> | <input type="text"/> | <input type="text"/> |
| 14 | <input type="text" value="3330014"/> | <input type="text"/> | <input type="text"/> |
| 15 | <input type="text" value="3330015"/> | <input type="text"/> | <input type="text"/> |
| 16 | <input type="text" value="3330016"/> | <input type="text"/> | <input type="text"/> |

← Phone number and Caller Name for each port

2. Choose **Static** as the *SIP Server Source*.
3. In the *Registrar Host* and *Proxy Host* fields, enter the address of the PC that hosts Asterisk.
4. In the *User Name* (equivalent to the phone number) column, enter a user name as defined in Asterisk.
5. You can also enter a *Friendly Name* for each port.
6. Click **Submit** to apply the changes.

SIP Authentication

The next step is to enter SIP authentication information for each port.

1. Click **SIP** and **Authentication** to go to the *SIP Authentication* web page.

You can enter up to 5 credentials for each port, but only one is needed in most cases.

Mediatix®

Device Info Management SIP Telephony Advanced

Configuration Interop Authentication Reboot

Authentication

| Unit Authentication | | | |
|---------------------|-------|----------|----------|
| Index | Realm | Username | Password |
| 1 | | | |
| 2 | | | |
| 3 | | | |
| 4 | | | |
| 5 | | | |

| User Agent Authentication | | | | |
|---------------------------|-------|----------|----------|----------|
| Port | Index | Realm | Username | Password |
| 1 | 1 | asterisk | 101 | num101 |
| 1 | 2 | | | |
| 1 | 3 | | | |
| 1 | 4 | | | |
| 1 | 5 | | | |
| 2 | 1 | asterisk | 102 | |
| 2 | 2 | | | |
| 2 | 3 | | | |
| 2 | 4 | | | |
| 2 | 5 | | | |
| 3 | 1 | | | |
| 3 | 2 | | | |
| 3 | 3 | | | |

Port 1

Port 2

Port 3

2. Type **asterisk** in the *Realm* field.
3. Enter the *Username* **101** and *Password* **num101** for port 1.
4. Click **Submit** to save the changes.

| | | | |
|----|---------|--|--|
| 22 | 3330022 | | |
| 23 | 3330023 | | |
| 24 | 3330024 | | |

SIP Registration

SIP Registration Command: noOp

Submit

Codec and DTMF Configuration

1. Click **Telephony** and **Codec** to go to the *Codec* page.

Codec and DTMF settings are configured on a port-by-port basis.

The screenshot shows the Mediatrix web interface. At the top, there are navigation tabs: Device Info, Management, SIP, **Telephony**, and Advanced. Below these are sub-tabs: Digit Maps, **CODEC**, Call Forward, Services, Misc, and Reboot. The 'CODEC' sub-tab is selected, and a dropdown menu on the left shows 'Port 1' selected. The main configuration area for 'Port 1' includes:

- Codec: g711-PCMU
- Jitter Buffer: Enable
- Number Buffer: 30
- Jitter Buffer: 125
- DTMF Transport: outOfBandUsingRtp
- Load Type: 101
- Port (same for all ports): 5004

Below this, there are sections for G.711 parameters:

- G.711 Law: Enable
- Law minimum packetization time: 10 ms
- Law maximum packetization time: 100 ms
- G.711 a-Law: Enable
- G.711 a-Law minimum packetization time: 10 ms
- G.711 a-Law maximum packetization time: 100 ms
- G.711 VAD: disable
- G.711 Comfort Noise Generation: disable

2. Set the *DTMF Transport* drop-down menu according to the DTMF transport mode you have defined in Asterisk.
 - If you have used *rfc2833*: set the parameter to **outOfBandUsingRtp**.
 - If you have used *info*, set the parameter to **outOfBandUsingSignalingProtocol**.
 - If you have used *inband*, set the parameter to **inBand**.

In this case, **OutOfBandUsingRTP** is used

3. Set the *Payload Type* field to **101**.
4. G.711 PCMU is the default codec.
5. Disable G.711 VAD (aka Silence Suppression):

You must turn off the Silence Suppression feature in the Mediatrix 41xx according to this Asterisk website: <http://www.voip-info.org/wiki/index.php?page=Asterisk+config+sip.conf> ,

Asterisk uses the incoming RTP Stream as a timing source for sending its outgoing Stream. If the incoming stream is interrupted due to silence suppression then musiconhold will be choppy. So in conclusion, you cannot use silence suppression. Make sure ALL SIP phones have disabled silence suppression. There is a solution for the silence suppression problem, see bug 5374 for details

6. Disable G711 Comfort Noise.
7. Enable (default) / Disable T.38 Fax if you are not using it in your VoIP setup.

8. Click **Submit** to apply the changes.

Done!

Reboot the gateway.

1. Click **Advanced** and **Reboot** to go to the *Reboot* page.



2. Click the **Reboot** button on the page that displays.



The changes you just made will become effective after the unit reboots.

After the unit comes back, observe that the *Ready* LED should now light up (or blink if not all the ports are configured and registered with the SIP server). You can then hook up a telephone and make some test calls with the Mediatrix 41xx.

Good Luck!

Advanced Settings

* Star Code and # Key Dial Map

By default, the Mediatrix 41xx dial map does not allow * and # keys. To do that, you must add the following dial map:

- **x.#** - for speed dialling, e.g. 267#, call will be dialled right away once the # key is hit
- ***xx** – for * code, e.g. *69
- **#xx** – for # code, e.g. #21

Here the first dial map is (*xx|#xx). The second dial map is x.# with # removed before the number is sent to Asterisk.

Mediatrix

Device Info Management SIP Telephony Advanced

Digit Maps CODEC Call Forward Services Misc Reboot

Digit Maps

General

Digit Map Timeout Inter Digit: 4000

Digit Map Timeout First Digit: 20000

Digit Map Timeout Completion: 60000

Allowed Digit Map

| Index | Activation | Digit Map | Remove Prefix | Add Prefix | Remove Suffix |
|-------|--|-----------|---------------|------------|---------------|
| 1 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | [*xx #xx] | 0 | | |
| 2 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.# | 0 | | # |
| 3 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 4 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 5 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 6 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 7 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 8 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 9 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |
| 10 | <input checked="" type="radio"/> Enable <input type="radio"/> Disable | x.T | 0 | | |

Call Transfer SIP Interop Setting for Asterisk 1.4

The SIP behaviour of Asterisk in the Call transfer scenario in version 1.4 is different than in previous Asterisk versions. The following MIB on the Mediatrix 41xx must be changed. To access this MIB parameter, the Mediatrix Unit Manager Network (UMN) is required. UMN can be downloaded from the Mediatrix Download Portal:

<https://support.mediatrix.com/DownloadPlus/Download.asp>

The MIB parameter is: *mediatrix.mediatrixExperimental.sipInteropMIB.sipInteropReplacesConfig*. This MIB has to be set to **useReplacesNoRequire**. The default value is “useReplacesWithRequire”:

Right Click at the unit
The Edit SNMP manual will pop up

1. Make sure Automatic GET box is checked

3. Hit the SET button

2. Select useReplacesNoRequire In the pull down menu

Details

| | | | |
|---------------|---|------|---|
| Type | Integer | Enum | 0: doNotUseReplaces 1: useReplacesWithRequire 2: useReplacesNoRequire |
| Status | Current | | |
| Default value | useReplacesWithRequire | | |
| Access | Read write | | |
| OID | 1.3.6.1.4.1.4935.99.20.1.5 | | |
| Description | Configures usage of the Replaces header mechanism used in a transfer. When supported by the target of the transfer, Replaces ensures a more seamless transfer by permitting the initiating party to effectively replace a current call by another instead of disconnecting the call to be replaced and creating a second call | | |

Get request result: SUCCESS
sipInteropTransferVersion [1.3.6.1.4.1.4935.99.20.1.10.0] = transfer0SUsingRefer02

Get request result: SUCCESS
sipInteropReplacesConfig [1.3.6.1.4.1.4935.99.20.1.5.0] = useReplacesWithRequire

T.38 Configuration

From version 1.4, Asterisk supports the negotiation and passthrough of T.38.

1. Modify the /etc/asterisk/udptl.conf configuration file so that it is as follow :

```
[general]
udptlstart=4000
udptlend=4999
T38FaxUdpEC = t38UDPRedundancy
T38FaxRateManagement = transferredTCF
T38FaxMaxDatagram = 400
udptlfecentries = 3
udptlfecspan = 3
```

2. Add the t38_udptl=yes to the “general” section of the /etc/asterisk/sip.conf

```
[general]
t38pt_udptl=yes
```

You can also add it to the SIP users and trunks supporting T.38:

```
[101]
type=friend
host=dynamic
nat=yes
qualify=yes
canreinvite=yes
dtmfmode=rfc2833
context=sip
username=101
secret=num101
t38pt_udptl=yes
```

By default, Asterisk supports speeds up to 9600. Mediatrix units are able to achieve speeds of up to 14400. To enable higher speeds, Asterisk needs to be recompiled. The following web page explains the procedure to enable higher speeds:

<http://www.voip-info.org/tiki-index.php?page=Asterisk%20T.38>

Depending on the Asterisk version, re-INVITES sent by the Mediatrix units to switch to T.38 might conflict with re- INVITES sent by the server. This happens especially when the “canreinvite” parameter is set to yes on the SIP user. If you experience problems with re- INVITES, it is recommended to disable the dataIfCngToneDetection variable using Unit Manager Network.

Syslog Server for Troubleshooting

For troubleshooting purpose, syslog of various levels can be enabled on the Mediatrix 41xx. In this example, Syslog messages are sent to the syslog server at 192.168.1.100, the syslog level is set to **Debug**.

The screenshot shows the Mediatrix web interface with the following configuration details:

| Syslog Info | |
|------------------------------|---------------|
| Syslog Configuration Source: | DHCP |
| Syslog Host: | 0.0.0.0 |
| Syslog Port: | 514 |
| Syslog Max. Severity: | Informational |

| Syslog Configuration | |
|------------------------------|---|
| Syslog Configuration Source: | <input checked="" type="radio"/> Static <input type="radio"/> DHCP |
| Static Syslog Host: | 192.168.1.100 |
| Static Syslog Port: | 514 |
| Syslog Max. Severity: | Debug |

Changes you have made require to reboot the device before they take effect.
Please click here to reboot the device.

Submit

References

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

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

<https://support.mediatrix.com/DownloadPlus/Download.asp>