



Enabling Live Communications at the Edge of IP Networks

Configuration Notes 286

Configuring Mediatrix 41xx FXS Gateway and 1204 FXO Gateway in a Remote Office Environment

June 22, 2011

Proprietary

© 2011 Media5 Corporation

Table of Contents

Introduction	3
About Mediatrix 41xx Series FXS Gateways	3
Application Scenario	4
Versions Supported	4
Mediatrix FXS (41xx) Units Configuration	5
Using the Web interface	5
Network Parameters Configuration.....	6
SIP Configuration	7
SIP Authentication	8
Codec and DTMF Configuration	9
Done!.....	11
Mediatrix FXO (1204) Unit Configuration	12
Using the Web interface	12
Network Parameters Configuration.....	13
SIP Configuration	14
Codec and DTMF Configuration	15
Done!.....	16
Advanced Settings.....	17
* Star Code and # Key Dial Map.....	17
Local 911 Call Using the Mediatrix Emergency Gateway Feature	18
Automatic Call Forwarding to an Auto-Attendant for all Inbound PSTN Calls via the 1204.....	19
Syslog Server for Troubleshooting	20
References	20

Introduction

This document outlines the configuration steps to set up Mediatrix 41xx series FXS gateway and the Mediatrix 1204 FXO gateways in a remote office environment. It assumes that you have a softswitch (or SIP server) properly installed in the network.

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:

- **4108** – Enterprise 8-port FXS gateway
- **4116** – Enterprise 16-port FXS gateway
- **4124** – Enterprise 24-port FXS gateway

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

- Superior Voice quality with dedicated DSP hardware
- Adaptive Jitter Buffer, G.168 Echo Cancellation
- T.38 Fax Relay support
- Easy-to-use web interface



Application Scenario

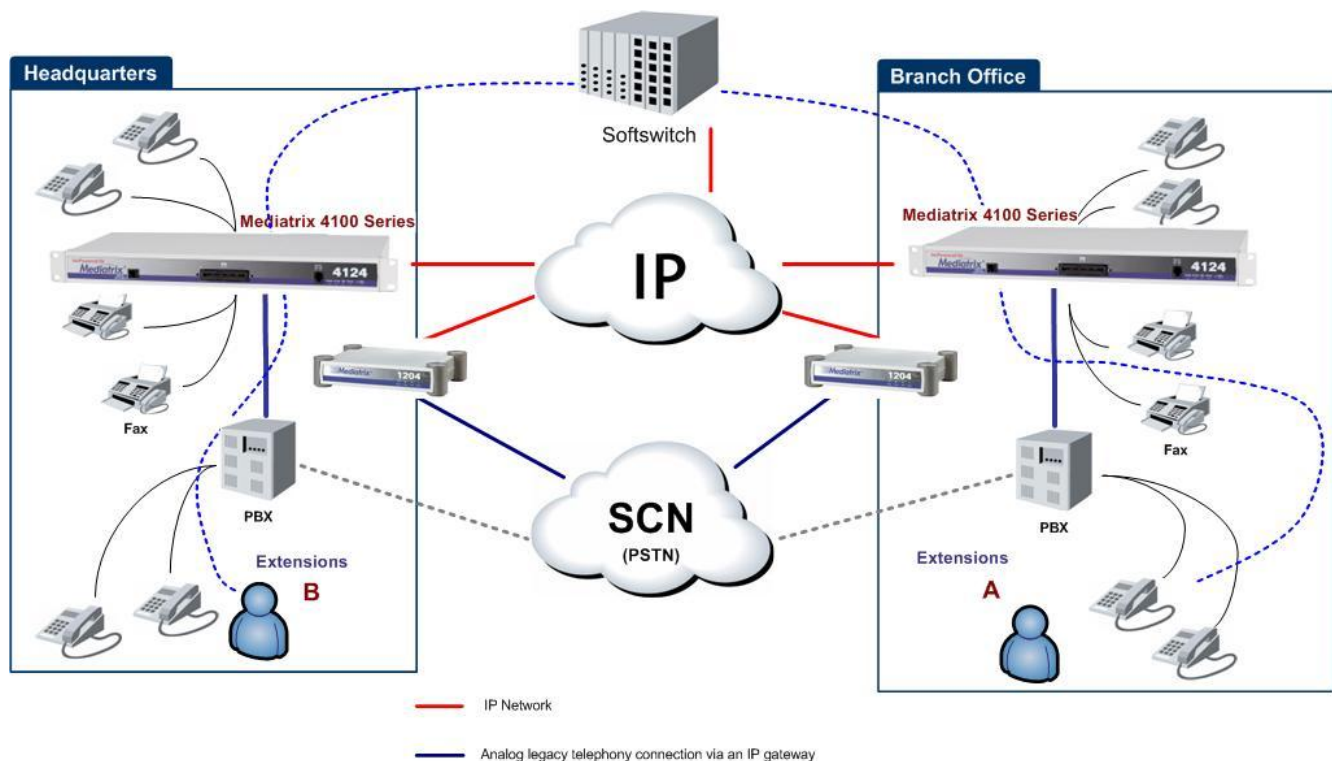
This is a typical deployment scenario of Mediatrix FXS and FXO gateway in a remote office VoIP environment. The gateways provide the following functionalities:

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

FXO gateway (1204) – PSTN access, local 911

Softswitch or IP PBX – 3rd Party device (or the Mediatrix IP Communication Server IPCS, which has lesser functionalities than a full-blown softswitch) that provides:

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



Versions Supported

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

- Mediatrix 1204: SIP 5.0.17.104
- Mediatrix 41xx: SIP 5.0.18.113

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

- Mediatrix 1204, 4108, 4116, and 4124 SIP Reference Manual
- Mediatrix 1204, 4108, 4116, and 4124 SIP Quick Start
- Mediatrix Unit Manager Network Administration Manual
- All relevant Softswitch Manuals

Mediatrrix FXS (41xx) Units Configuration

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

<https://support.mediatrrix.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 Mediatrrix reseller for more details.

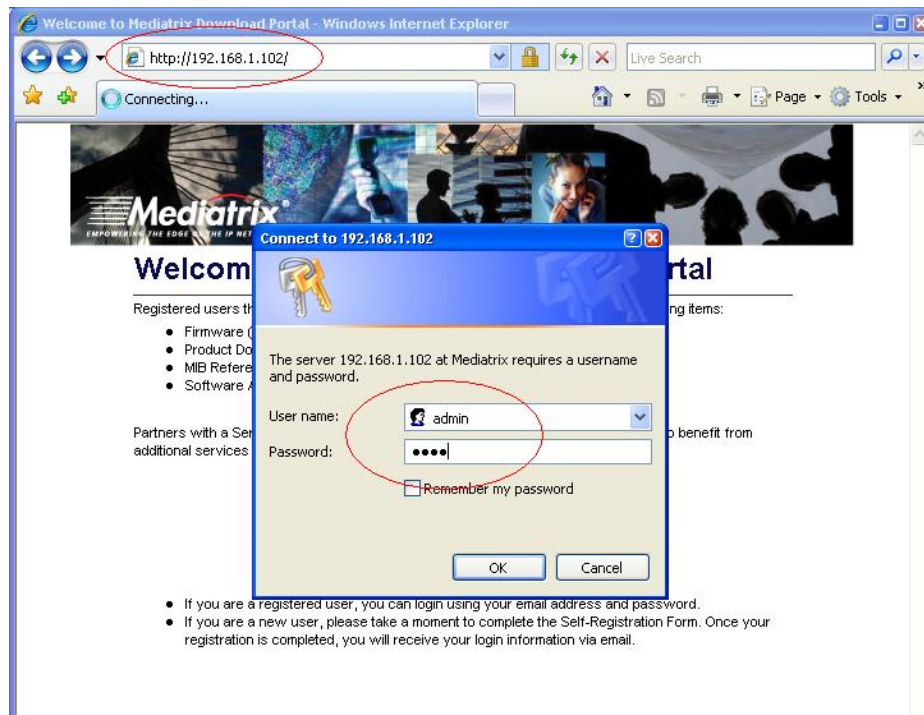
Using the Web interface

The web interface may be used to:

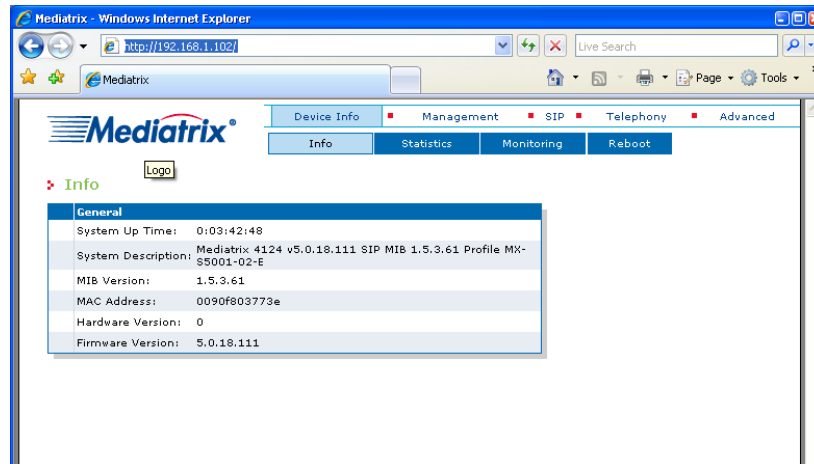
- View the status of the Mediatrrix 41xx.
- Set numerous parameters of the Mediatrrix 41xx.

Mediatrrix 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:

1. In the web browser's address field, type the IP address of the Mediatrrix 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**.



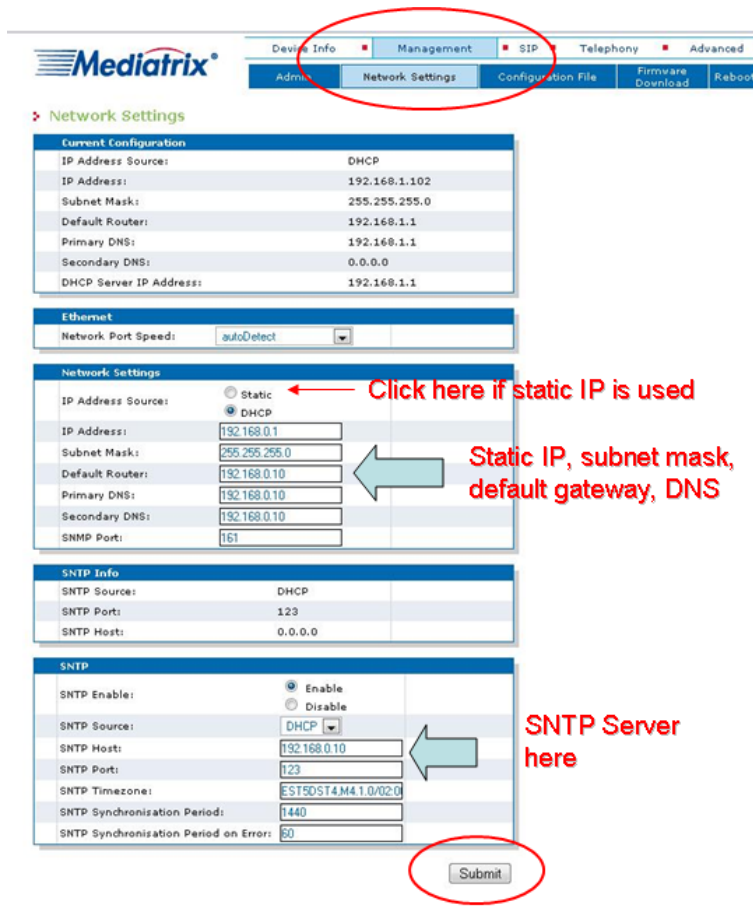
- The system information screen then appears, giving information about the firmware version, hardware revision, system up-time and MAC address.



Network Parameters Configuration

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

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



- 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

The SIP Configuration usually includes two portions:

- Information about the signalling servers, e.g., SIP proxy, SIP registrar, port number, etc.
- SIP account setting (aka phone number) for each port; the account should be configured on the softswitch first

The following are the steps to configure the SIP parameters:

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 ☐ DHCP
SIP Port:	0
Registrar Host:	10.1.119.229
Registrar Port:	0
Proxy Host:	10.1.119.229
Proxy Port:	0
Outbound Proxy Host:	0.0.0.0
Outbound Proxy Port:	0
Unregistered Port Behavior:	disable port

SIP User Configuration

Port	User Name	Friendly Name	Other Accepted Username
1	101	John	
2	102	David	
3	3330003		
4	3330004		
5	3330005		
6	3330006		
7	3330007		
8	3330008		
9	3330009		
10	3330010		
11	3330011		
12	3330012		
13	3330013		
14	3330014		
15	3330015		
16	3330016		

2. Choose **Static** as the *SIP Server Source*.

3. In the *Registrar Host* and *Proxy Host* fields, enter the address of the softswitch signalling server.
4. In the *User Name* (equivalent to the phone number) column, enter a user name as defined in the softswitch. The port will use this as the username to register with the softswitch.
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
1		
2		
3		
4		
5		

Port	Index	Realm	Username	Password
1	1	asterisk	101	num101
1	2			
1	3			
1	4			
1	5			
2	1	asterisk	102	num102
2	2			
2	3			
2	4			
2	5			
3	1			
3	2			
3	3			

SIP Registration

SIP Registration Command: noOp

Submit

Port 1

Port 2

Port 3

2. Type the proper information the *Realm* field; the Realm can be found from the softswitch vendor. In this example, enter **asterisk**.
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 Mediatix web interface. At the top, there are tabs for 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. On the left, a dropdown menu for 'Port 1' is open, showing a list of ports from Port 1 to Port 20. The configuration fields for Port 1 are displayed on the right. The 'Codec' is set to 'g711-PCMU'. The 'Jitter Buffer' is set to 'Enable'. The 'Transport' is set to 'outOfBandUsingRtp'. The 'Load Type' is set to '101'. The 'Port (same for all ports)' is set to '5004'. Below these, there are sections for 'G.711 a-Law' and 'G.711 VAD' parameters, all of which are set to 'disable' or '100 ms'.

2. Set the *DTMF Transport* drop-down menu according to the DTMF transport mode you have defined in the softswitch.
 - o If you are using *rfc2833*: set the parameter to **outOfBandUsingRtp**.
 - o If you are using *SIP Info*, set the parameter to **outOfBandUsingSignalingProtocol**.
 - o If you are sending *DTMF inband*, set the parameter to **inBand** (default). Inband is not recommended if you are intending to use a codec other than G.711.

In this case, **OutOfBandUsingRTP** is used.

Mediatrix

Device Info Management SIP Telephony Advanced

Digit Maps CODEC Call Forward Services Misc Reboot

CODEC

Port 1

General

Preferred Codec: g711-PCMU

Adaptive Jitter Buffer: ☒ Enable ☐ Disable

Target Jitter Buffer: 30

Maximum Jitter Buffer: 125

DTMF Transport: outOfBandUsingRtp

DTMF Payload Type: 101

RTP Base Port (same for all ports): 5004

Voice - G.711

G.711 u-Law: ☒ Enable ☐ Disable

G.711 u-Law minimum packetization time: 10 ms

G.711 u-Law maximum packetization time: 100 ms

G.711 a-Law: ☒ Enable ☐ Disable

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

Voice - G.729

G.729: ☒ Enable ☐ Disable

G.729 minimum packetization time: 10 ms

G.729 maximum packetization time: 100 ms

G.729 VAD: ☒ Enable ☐ Disable

OutOfBandUsingRTP = rfc2833
Payload number = 101

VAD = Disable
Comfort Noise = Disable

- Set the *Payload type* field to match your setting in the softswitch and other VoIP devices. In this example, the payload type **101** is used.
- Set your preferred codec. G.711 PCMU is the default codec.
- Set your codec-specific parameters, e.g., Comfort noise, Silence Suppression/Voice Activity Detection (VAD), packetization time, etc.
- Enable (default) / Disable T.38 Fax if you are not using it in your VoIP setup.

Fax

T.38: ☒ Enable ☐ Disable

T.38 Protection Level: 3

T.38 Final Frames Redundancy: 0

Preferred Clear Channel Codec: g711-PCMU

Submit

- 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.

Mediatrrix FXO (1204) Unit Configuration

Similar to the 41xx series, the Mediatrrix 1204 FXO gateway also features an embedded Web server. Most of the commonly used parameters are accessible from the web interface. The Mediatrrix Unit Manager Network (UMN) software is needed if access to full unit configuration is required. The UMN can be downloaded from the Mediatrrix Portal:

<https://support.mediatrrix.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 Mediatrrix reseller for more details.

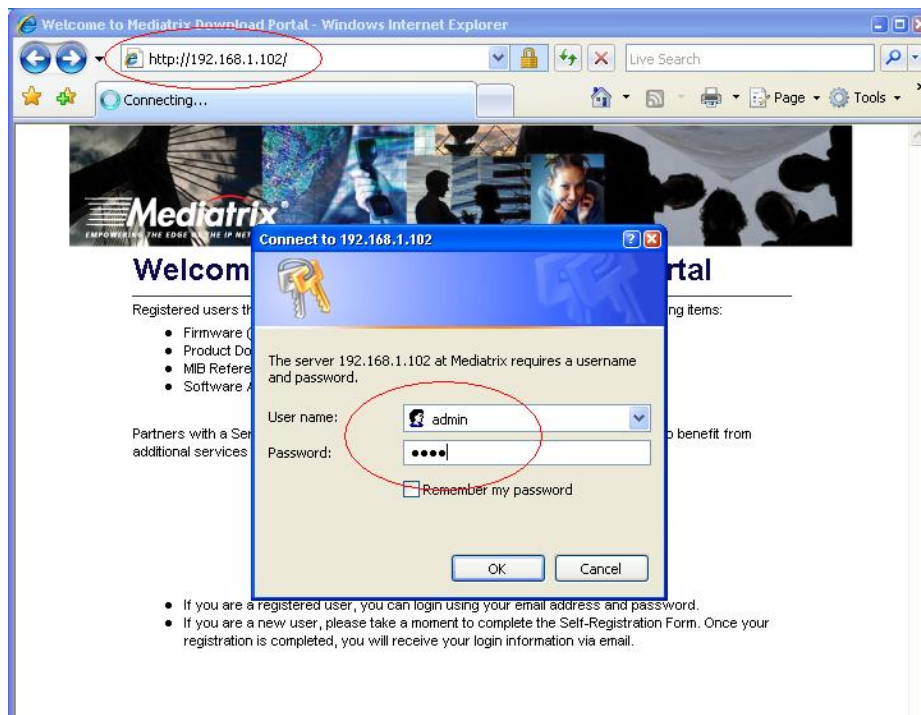
Using the Web interface

The web interface may be used to:

- View the status of the Mediatrrix 1204.
- Set numerous parameters of the Mediatrrix 1204.

Mediatrrix 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:

1. In the web browser's address field, type the IP address of the Mediatrrix 1204 (if you have performed a recovery mode, this is 192.168.0.1).
2. Enter the default login name **admin** and password **1234**.



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

General	
System Up Time:	0:00:04:00
System Description:	Mediatrix 1204 v5.0.17.104 SIP MIB 1.5.3.61 Profile MX-S5001-02-E
MIB Version:	1.5.3.61
MAC Address:	0090f8005a64
Hardware Version:	8
Firmware Version:	5.0.17.104

Network Parameters Configuration

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

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

Network Settings

Current Configuration

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
DHCP Server IP Address:	192.168.1.1

Ethernet

Network Port Speed: autoDetect

Network Settings

IP Address Source: ☐ Static ☒ DHCP

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

SNMP Port: 161

SNTP Info

SNTP Source:	DHCP
SNTP Port:	123
SNTP Host:	0.0.0.0

SNTP

SNTP Enable: ☒ Enable ☐ Disable

SNTP Source: DHCP

SNTP Host: 192.168.0.10

SNTP Port: 123

SNTP Timezone: EST/SDT/M4.1.0/02.0

SNTP Synchronization Period: 1440

SNTP Synchronization Period on Error: 60

Submit

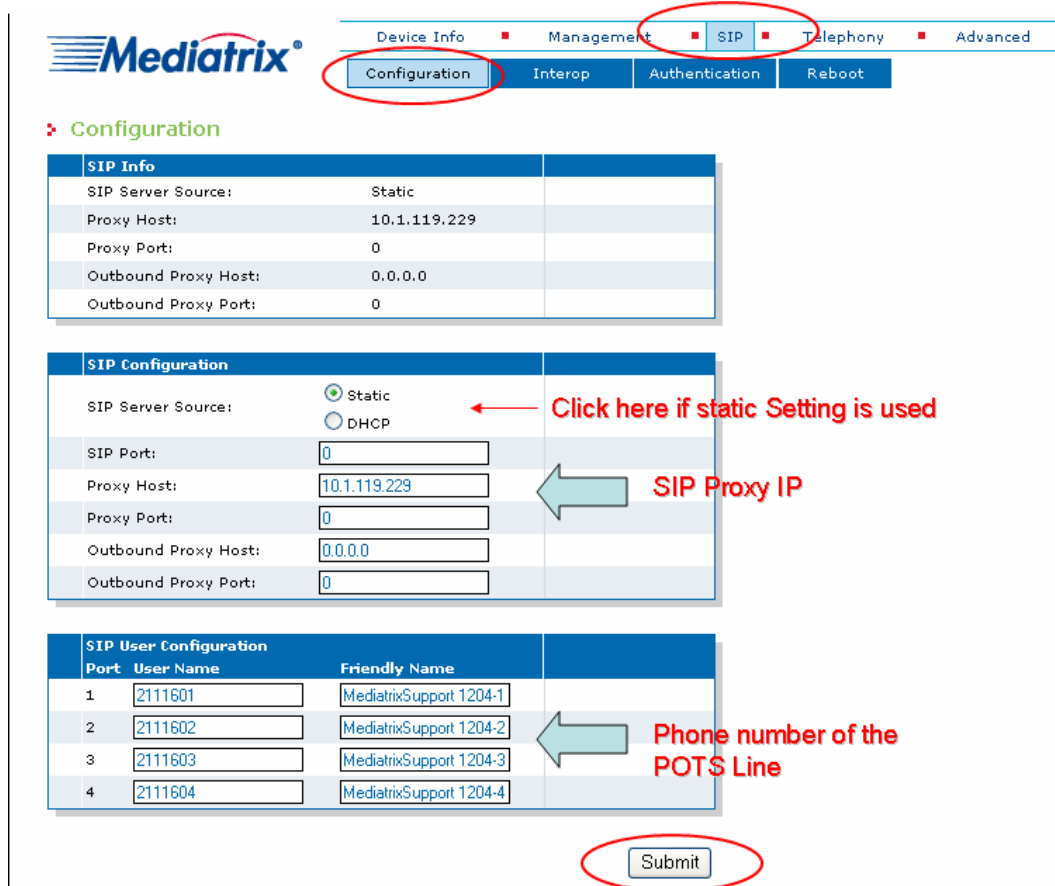
- In the *IP Address Source* option, select **DHCP** or **Static**.
DHCP is the default selection.
- If you are using a Static IP address, select **Static** and enter the IP address, subnet mask, default router and DNS server IP in the proper fields.

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

The SIP Configuration usually includes information about the signalling servers, e.g. SIP proxy, port number, etc. Note that unlike the 41xx FXS gateway, SIP Registration is not needed for the 1204.

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



Configuration

SIP Info	
SIP Server Source:	Static
Proxy Host:	10.1.119.229
Proxy Port:	0
Outbound Proxy Host:	0.0.0.0
Outbound Proxy Port:	0

SIP Configuration	
SIP Server Source:	☒ Static ☐ DHCP
SIP Port:	0
Proxy Host:	10.1.119.229
Proxy Port:	0
Outbound Proxy Host:	0.0.0.0
Outbound Proxy Port:	0

SIP User Configuration		
Port	User Name	Friendly Name
1	2111601	MediatixSupport 1204-1
2	2111602	MediatixSupport 1204-2
3	2111603	MediatixSupport 1204-3
4	2111604	MediatixSupport 1204-4

Submit

2. In the *SIP Configuration* section, choose **Static** as the *SIP Server Source*.
3. In the *Proxy Host* field, enter the address of the softswitch signalling server.
4. In the *User Name* (equivalent to the phone number) column, enter the phone number of the POTS line to which the port is connected. This is optional.
5. Click **Submit** to save the changes.

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.

CODEC

Port 1
Port 2
Port 3
Port 4

Codec: g711-PCMU

Adaptive Jitter Buffer: ☒ Enable ☐ Disable

Target Jitter Buffer: 30

Maximum Jitter Buffer: 125

DTMF Transport: outOfBandUsingRtp

DTMF Payload Type: 101

RTP Base Port (same for all ports): 5004

Voice - G.711

G.711 u-Law: ☒ Enable ☐ Disable

G.711 u-Law minimum packetization time: 10 ms

G.711 u-Law maximum packetization time: 100 ms

G.711 a-Law: ☒ Enable ☐ Disable

G.711 a-Law minimum packetization time: 10 ms

G.711 a-Law maximum packetization time: 100 ms

G.711 VAD: conservative

G.711 Comfort Noise Generation: customNoise

2. Set the *DTMF Transport* drop-down menu according to the DTMF transport mode you have defined in the softswitch:
 - o If you are using *rfc2833*, set the parameter to **outOfBandUsingRtp**.
 - o If you are using *SIP Info*, set the parameter to **outOfBandUsingSignalingProtocol**.
 - o If you are sending DTMF *inband*, set the parameter to **inBand** (default). Inband is not recommended if you are intending to use codec other than G.711.

In this example, **OutOfBandUsingRTP** is used

3. Set the *Payload Type* field to match your setting in the softswitch and other VoIP devices. In this example, the payload type **101** is used.
4. Set your preferred codec. G.711 PCMU is the default codec.
5. Set your codec-specific parameters, e.g., Comfort noise, Silence Suppression/Voice Activity Detection (VAD), packetization time, etc.

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

7. 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, you can then hook up the PSTN lines and make some test calls with the Mediatrix FXO 1204 gateway. When you dial into the 1204 from the PSTN, you'll get a second dial tone. You can then dial any IP extension numbers to reach internal phones. If you want to forward all inbound calls into an auto-attendant, please check out **Automatic Call Forwarding** in the *Advanced Setting* section.

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.



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	☒ Enable ☐ Disable	(*xx #xx)	0		
2	☒ Enable ☐ Disable	x.#	0		#
3	☒ Enable ☐ Disable	x.T	0		
4	☒ Enable ☐ Disable	x.T	0		
5	☒ Enable ☐ Disable	x.T	0		
6	☒ Enable ☐ Disable	x.T	0		
7	☒ Enable ☐ Disable	x.T	0		
8	☒ Enable ☐ Disable	x.T	0		
9	☒ Enable ☐ Disable	x.T	0		
10	☒ Enable ☐ Disable	x.T	0		

Local 911 Call Using the Mediatrix Emergency Gateway Feature

The Emergency Call service (also called urgent gateway) allows a “911”-style service on the Mediatrix 41xx. It allows a user to dial a special digit map resulting in a message being sent to a specified urgent gateway (the Mediatrix 1204 in this case), bypassing the softswitch and any other intermediaries.

If enabled, whenever the user dials the specified digit map, a message is sent to the target address.

In this example, when the user dials **911** or ***60**, a 911call is dialled out of the PSTN gateway (a Mediatrix 1204 with IP address 192.168.1.102). All the configuration is done on the 41xx only:

1. Click **Advanced** and select **Emergency**.
2. Set the *Emergency Call Activation* choice to **Enable**.
3. Enter the *Emergency Call Target*. This is the destination SIP URI; it is in the format **sip:911@192.168.1.102**, with 911 being the number intended to be dialled out to the PSTN, and 192.168.1.102 being the local PSTN gateway.
4. Enter the *Emergency Call Digit Map*. These are the digits the user dials in order to trigger the emergency call; in this example, the user can either dial 911 or *60 to call emergency. If only 911 is needed, just put **911**.
5. Click **Submit** to complete the process.

Mediatrix®

Device Info Management SIP Telephony **Advanced**

QoS **Emergency** STUN Reboot

✚ Emergency

Emergency Call	
Emergency Call Activation:	☒ Enable ☐ Disable
Emergency Call Target:	sip:911@192.168.1.102
Emergency Call Digit Map:	[911*60]

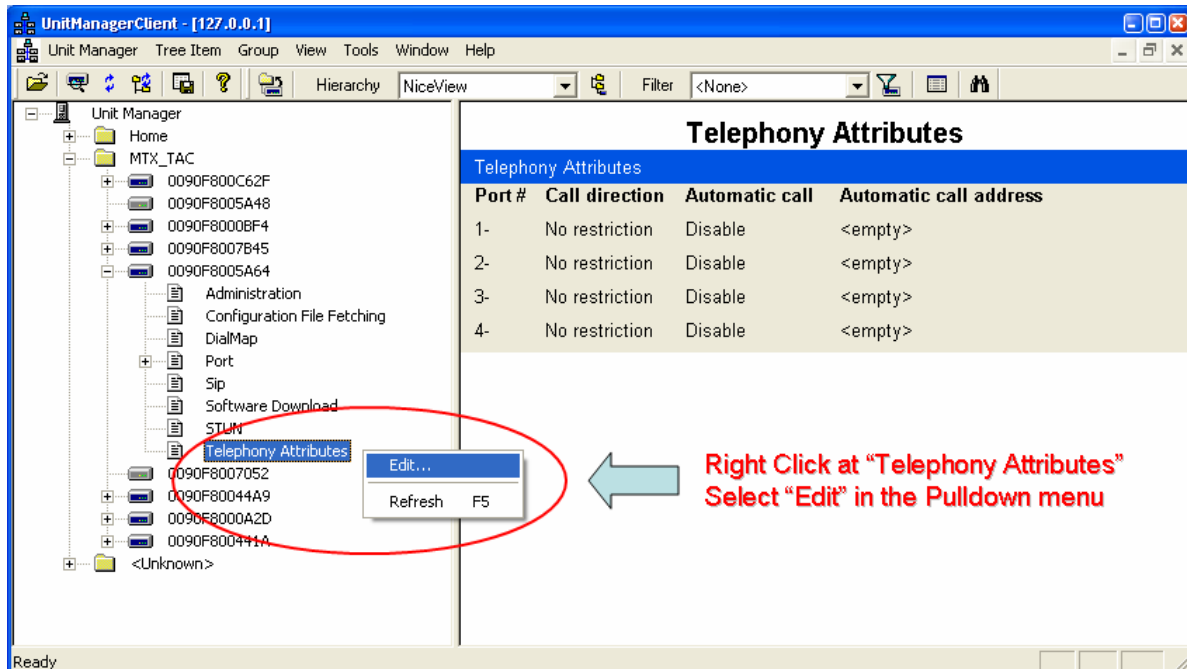
Submit

Set Activation to "Enable"
Enter URI for Emergency Call
Enter Digit Map that will trigger the call

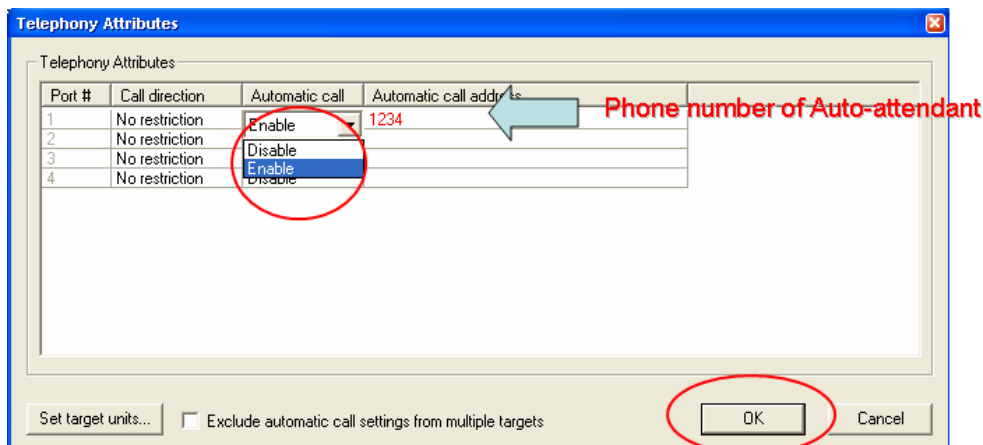
Automatic Call Forwarding to an Auto-Attendant for all Inbound PSTN Calls via the 1204

You'll need the Mediatrix Unit Manager Network in order to access the Automatic Call feature on the 1204. The Automatic Call function allows all inbound PSTN calls to the 1204 being automatically forwarded to a pre-defined number. Without this being set, the caller will hear a second dial tone instead.

1. In the UMN, right-click **Telephony Attributes** and select **Edit**.



2. The *Telephony Attributes* window pops up. Set the *Automatic Call* field to **Enable** and put your auto-attendant phone number in the *Automatic Call Address* field. Click **OK** to complete the process.



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 'Monitoring' tab selected. The 'Syslog Info' section displays the current configuration: Syslog Configuration Source is DHCP, Syslog Host is 0.0.0.0, Syslog Port is 514, and Syslog Max. Severity is Informational. The 'Syslog Configuration' section shows the configuration source set to Static, with the Static Syslog Host set to 192.168.1.100 and the Static Syslog Port set to 514. The Syslog Max. Severity dropdown menu is open, showing options: Disabled, Critical, Error, Warning, Informational, and Debug. The 'Submit' button is visible. A red message at the bottom states: 'Changes you have made require to reboot the device before they take effect. Please click here to reboot the device.'

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

Syslog Configuration	
Syslog Configuration Source:	☒ Static ☐ DHCP
Static Syslog Host:	192.168.1.100
Static Syslog Port:	514
Syslog Max. Severity:	Informational Disabled Critical Error Warning Informational Debug

Submit

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

References

For more information on the Mediatrix products, visit these links:

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