

AudioCodes MP202 FXS gateway

Rainbow Hub provisioning guide

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MP202 version: 4.4.9_build_144

Document revisions

Revised on	Version	Description	Revised by
01.09.2020	1.0	Initial version	YM
26.05.2021	1.1	Adopted to reflect TLS 1.2 and SIP proxy requirements	MC
12.12.2021	1.2	Added certificate upload section; refresh with updated screenshots	YM
26.05.2023	1.3	Refresh certificate upload section	CM

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1. General

This document provides overview of AudioCodes MP202 FXS VoIP gateway (ATA), when used for connection of analog device to Rainbow Hub.

Configuration on Rainbow Hub side (user creation, license allocation, obtaining of SIP account password and domain) is not covered by this document.

2. Rainbow integration parameters

There is a number of provisioning parameters related to Rainbow that need to be configured in the MP202 device. Below information must be obtained per each connected analog device from Rainbow Company Administration interface - under *Communication – Devices – SIP options* prior to MP202 configuration:

- **SIP registrar domain name** (e.g. *75.eu1.sip.openrainbow.com*) and port (e.g. *5061*)
- **SIP proxy domain name** (e.g. *lb02.eu1.sip.openrainbow.com*) and port (e.g. *5061*)
- **SIP User name** (e.g. *105*) – usually equals to extension number
- **SIP Password**
- **CA certificate chain** (download a *.pem file)

Note: you may need to split the downloaded certificate *.pem file into three separate certificates, using any text editor (e.g. Notepad).

Rainbow SIP certificates are signed by an official authority (The USERTRUST Network) that might be already trusted by some equipment. In that case there is no need to upload any certificates. Rainbow certificates are already accepted by the equipment firmware.

Only the third certificate (the Root CA) of the chain is mandatory and must be uploaded if not already embedded in the equipment; It expires on 19th January 2038 (fingerprint HA-256: E7 93 C9 B0 2F D8 AA 13 E2 1C 31 22 8A CC B0 81 19 64 3B 74 9C 89 89 64 B1 74 6D 46 C3 D4 CB D2).

Device information

Information

SIP options

Member

SIP account settings

Domain 75.eu1.sip.openrainbow.com

User name 105

Password



Port 5061

Transport protocol TLS

CA certificates chain

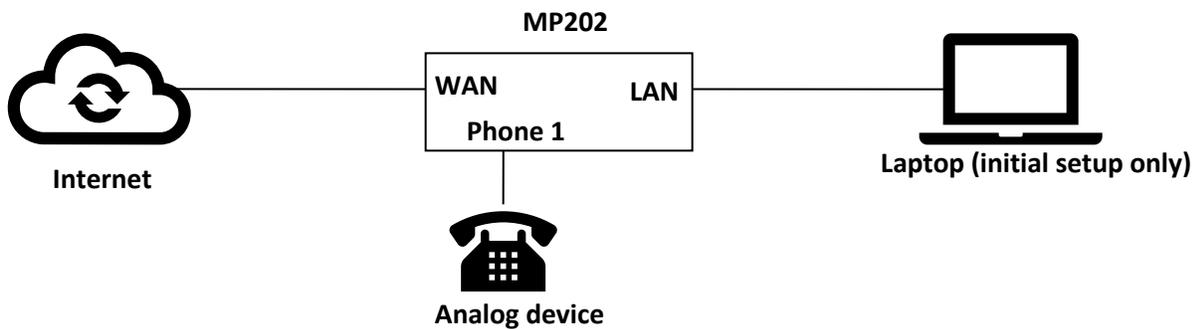
Download



3. IP address configuration

MP202 can provide, in addition to FXS to SIP gateway function, multiple additional functions: firewall, NAT, routing, etc. For the purpose of pure FXS SIP gateway, those functions are not required. To simplify further management of MP202, there are few changes that should be done to the configuration.

Typically, the MP202 will be connected to customer network via WAN Ethernet interface, and LAN interface will be not utilized. However, initial configuration of the MP202 must be done by a PC, connected to the LAN interface. The PC LAN adapter needs to be configured with Automatic IP address assignment – DHCP (MP202 runs DHCP server on LAN interface).



- Connect **WAN** port of MP202 to customer's LAN
- Connect laptop ethernet port to MP202 **LAN** port
- On your PC, verify that the Local Area Connection status is "Connected", by clicking Start > Settings > Network Connections, and then double-clicking the Local Area Connection icon.
- If the LAN status is "Disconnected", click Properties, select 'Internet Protocol (TCP/IP)', and then click Properties; ensure that the 'Obtain an IP address automatically' option is selected.
- Open a Web browser and enter the URL address <http://MP20x.home> (or <http://MP202.home>)
- Default username of MP202 is admin; default password is admin. It is recommended to change the password and write it down
- Go to Network Connections menu, and click “edit” icon on the right of WAN Ethernet interface:



- By default, WAN interface obtains its IP address, mask, default gateway and DNS address from customer's DHCP server. Verify that the parameters obtained from DHCP are correct.

WAN Ethernet Properties

General Settings Routing Port Advanced

Name:	WAN Ethernet
Device Name:	eth1
Status:	Connected
Network:	WAN
Connection Type:	Ethernet
Download Rate:	100Mbps
Upload Rate:	100Mbps
MAC Address:	00:90:8f:b6:f3:87
IP Address:	192.168.100.120
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.100.254
DNS Server:	8.8.8.8
IP Address Distribution:	Disabled
Received Packets:	985794
Sent Packets:	23759
Time Span:	75:42:11

- If needed, the IP parameters can be set statically by going to Settings menu. After configuring all parameters, click Apply:

WAN Ethernet Properties

General Settings Routing Port Advanced

Device Name:	eth1
Status:	Connected
Schedule:	Always
Network:	WAN
Connection Type:	Ethernet
Physical Address:	00:90:8f:b6:f3:87
MTU:	Automatic 1500
Internet Protocol:	Use the Following IP Address
IP Address:	192.168.101.225
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.101.254
DNS Server:	Use the Following DNS Server Addresses
Primary DNS Server:	8.8.8.8
Secondary DNS Server:	0.0.0.0
IP Address Distribution:	Disabled

OK Apply Cancel

- Go to “Advanced” top menu, and disable Internet Connection Firewall. This will allow you to connect to the MP202 via WAN interface IP address; you will not need the LAN port anymore

WAN Ethernet Properties

General Settings Routing Port **Advanced**

Preferred for VoIP Enabled

Preferred for CWMP Enabled

Internet Connection Firewall Enabled

Additional IP Addresses

IP Address	Subnet Mask	Action
New IP Address		

OK Apply Cancel

- Click “Apply” to save the settings
- Go to Advanced / Diagnostics and check connectivity to *eu1.sip.openrainbow.com*. Both resolving and ping should work

Advanced

About MP20x Certificates Configuration File DNS Server **Diagnostics** Feature Key Firmware Upgrade

IP Address Distribution LLDP Network Objects PPPoE Relay Personal Domain Name (Dynamic DNS) Protocols Reboot

Regional Settings Remote Administration Restore Factory Settings Routing SSH Scheduler Simple Network Management Protocol (SNMP)

System Settings Time Settings Universal Plug and Play Users

Diagnostics

Diagnostics Debug

Ping (ICMP Echo)

Destination:

Number of pings:

Status: **Test succeeded**

Packets: 4/4 transmitted, 4/4 received, 0% loss

Round Trip Time: Minimum = 84 ms
Maximum = 93 ms
Average = 87 ms

ARP

Destination:

3. Voice over IP configuration

- Go to “Voice over IP” main / Signaling Protocol” menu, click on “Advanced” at the bottom and configure TLS as “SIP Transport Protocol”, and “Gateway name – User domain value”. Use domain value obtained from Rainbow Provisioning (e.g. *75.eu1.sip.openrainbow.com*):

The screenshot shows the 'Voice Over IP' configuration interface. On the left is a navigation menu with options: Home, Quick Setup, Network Connections, Security, Voice Over IP, QoS, Advanced, System Monitoring, and Logout. The main area has a top navigation bar with tabs: Signaling Protocol, Dialing, Media Streaming, Voice and Fax, Services, Line Settings, Extension Settings, Speed Dial, and Telephone Interface. The 'Signaling Protocol' section is active and contains the following fields:

Signaling Protocol:	SIP
SIP Transport Protocol:	TLS
Local SIP Port:	5060
Local SIP TLS Port:	5061
<input type="checkbox"/> Use sips on TLS	
Gateway Name - User Domain:	75.eu1.sip.openrainbow.com
<input checked="" type="checkbox"/> Enable PRACK	
<input checked="" type="checkbox"/> Include ptime in SDP	
<input checked="" type="checkbox"/> Enable rport	
<input type="checkbox"/> Connect media on 180	

- Enable “Use SIP Proxy” and “Use SIP Proxy IP and Port for Registration”. Configure “Host name” and “Proxy Port” with values obtained from Rainbow Provisioning (e.g. *lb02.eu1.sip.openrainbow.com* port *5061*):

The screenshot shows the 'SIP Proxy and Registrar' configuration section. It contains the following fields:

<input checked="" type="checkbox"/> Use SIP Proxy	
Host Name or Address:	lb02.eu1.sip.openrainbow.com
Proxy Port:	5061
Maximum Number of Authentication Retries:	4
<input checked="" type="checkbox"/> Use SIP Proxy IP and Port for Registration	
Register Expires:	3600 Seconds
Register Failed Expires:	60 Seconds
Sip Security:	Allow All SIP traffic
Redundancy Mode:	None
<input type="checkbox"/> Enable Keep Alive	
<input type="checkbox"/> Use SIP Outbound Proxy	

- Click “Apply” to save the changes
- Go to “Dialing” top menu, click on “Advanced” and enable Inband DTMF:

Home Quick Setup Network Connections Security Voice Over IP QoS Advanced System Monitoring Logout

Signaling Protocol **Dialing** Media Streaming Voice and Fax Services Line Settings Extension Settings Speed Dial Telephone Interface

Dialing Parameters

Dialing Timeout: Seconds

Interdigit Short Timeout: Seconds

Phone Number Size: Digits

Enabled dialing complete key

Complete dialing key:

Dial Tone Timeout: Seconds

Reorder tone timeout: Seconds

Unanswered call timeout: Seconds

Howler tone timeout: Seconds

Flash min: milliseconds

Flash max: milliseconds

Enable Re-Answer Timeout

Send DTMF Out-Of-Band:

Digit Map:

Dial Plan:

Key Sequence

Flash keys sequence style:

OK Apply Cancel Basic <<

- Click “Apply” to save the changes
- Go to “Media Streaming” top menu and change default 1st codec from u_Law to A-Law; enable SRTP:

Home Quick Setup Network Connections Security Voice Over IP QoS Advanced System Monitoring Logout

Signaling Protocol Dialing **Media Streaming** Voice and Fax Services Line Settings Extension Settings Speed Dial Telephone Interface

Media Streaming Parameters

Local RTP Port Range - Contiguous Series of 8 Ports Starting From:

DTMF Relay RFC2833 Payload Type (default value 101):

G.726/16 Payload Type (default value 98):

Quality of Service Parameters

Type Of Service (Hex):

Wide-Band Restrictions

Enabled

SRTP

Enabled

Method:

Codecs

Codecs Priority	Supported Codecs	Packetization Time (milliseconds)
1st Codec	<input type="text" value="G.711, 64kbps, A-Law"/>	<input type="text" value="20"/>
2nd Codec	<input type="text" value="G.711, 64kbps, A-Law"/>	<input type="text" value="20"/>
3rd Codec	<input type="text" value="G.729, 8kbps"/>	<input type="text" value="20"/>
4th Codec	<input type="text" value="G.726, 16kbps"/>	<input type="text" value="20"/>
5th Codec	<input type="text" value="G.726-32, 32kbps"/>	<input type="text" value="20"/>

OK Apply Cancel Basic <<

- Click “Apply” to save the changes
- Go to “Voice and Fax” top menu and configure below parameters. Click “Apply” when finished.

Media Streaming **Voice and Fax** Services Line Settings Extension Settings Speed Dial Telephone Interface

Gain Control

Enable Automatic Gain Control

Jitter Buffer

Minimum Delay (10 to 150 milliseconds): milliseconds

Optimization Factor (1 to 13):

Silence Compression

Enable Silence Compression

Echo Cancellation

Enable Echo Cancellation

Artificial Bandwidth Extension

Enable Artificial Bandwidth Extension

Fax and Modem Settings

Fax Transport Mode:

T38 Version:

Max Rate:

Max Buffer:

Max Datagram:

Error Correction Mode

Image Data Redundancy Level:

T30 Control Data Redundancy Level:

Fax Relay Jitter Buffer Delay:

Modem Transport Mode:

Modem Bypass Payload Type:

Fax/Modem Bypass Codec:

CED Transfer Mode:

CNG Transfer Mode:

- Go to “Line Settings”, and click edit icon to configure SIP parameters for each analog device, connected to the MP202 phone port (SIP parameters should be obtained from Rainbow Provisioning)
 - “User ID” – should be extension number (e.g.105)
 - “Authentication user name” – **SIP user name**; should be in format of <User name>@<SIP Domain name> e.g. 105@75.eu1.sip.openrainbow.com
 - “Authentication password” – **SIP password** as obtained from Rainbow provisioning

Home Quick Setup Network Connections Security Voice Over IP QoS Advanced System Monitoring Logout

Signaling Protocol Dialing Media Streaming Voice and Fax Services **Line Settings** Extension Settings Speed Dial Telephone Interface

Voice Over IP

Line	User ID	Display Name	Action
<input checked="" type="checkbox"/> 1	105	Line 1	
<input type="checkbox"/> 2	1975	pancode_shahar	

Line	User ID	Phone2
1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

OK Apply Cancel



The image shows a web-based configuration interface for a phone system. On the left is a dark blue sidebar with a menu containing: Home, Quick Setup, Network Connections, Security, Voice Over IP, QoS, Advanced, System Monitoring, and Logout. The main area is titled "Line Settings" and features a phone icon. It contains several configuration sections: "Line Number" (1), "User ID" (105), "Block Caller ID" (unchecked), "Display Name" (Line 1), and "Extensions Registered" (105, Phone2). The "SIP Proxy" section includes "Authentication User Name" (105@75.eu1.sp.openrainbc) and "Authentication Password" (masked with asterisks). The "Advanced Line Parameters" section has a checked checkbox for "Enable Supplementary Services". At the bottom are three buttons: "OK", "Cancel", and "Basic <<".

Line Number:	1
User ID:	105
<input type="checkbox"/> Block Caller ID	
Display Name:	Line 1
Extensions Registered:	105, Phone2
SIP Proxy	
Authentication User Name:	105@75.eu1.sp.openrainbc
Authentication Password:	*****
Advanced Line Parameters	
<input checked="" type="checkbox"/> Enable Supplementary Services	

OK Cancel Basic <<

- Click "OK" after configuring above parameters, and "Apply" on "Line settings" screen

4. CA Certificate upload

- Go to Advanced left panel menu – Certificated
- Click on “CA” tab



- Click on “Upload Certificate” and choose the third part of the CA certificates file downloaded from Rainbow provisioning. Verify that certificate is uploaded correctly.
 - Note: the pem file downloaded from Rainbow contains all the certificates chain used by Rainbow. Only the third one must be uploaded, you need to use any text editor to split the downloaded *.pem certificate file into three separate files, and upload it:

