SOLIDYNE
DIGITAL POWER

Wi Fi

USB

BLUETOOTH

MX 2200

portable mixer & IP codec

OWNER'S MANUAL

Índice de contenido

About this manual

review, June 2024

Packing list

When unpacking the unit, it is advisable to conduct a visual inspection to ensure that the equipment has not been damaged during transportation. Along with the console, you will receive the following items:

- *Power supply for continuous use or battery charge: 95 – 240 V 50/60 Hz input / 12 V - 1.25 A output*
- *1 USB ANTENNA WI-FI 2.4/5 GHz*
- *1 USB type AB cable*
- *Ethernet cable RJ45 1,2 meters*
- *Printed user manual*
- *Guarantee card*

1 Features

Solidyne MX-2200 is a portable mixer-console and IP audio CODEC for live remotes. It's an all in one device that manages the audio peripherals to generate the program signal and multiple wired and wireless technologies.

MX-2200 operation is simple and intuitive. The CODEC settings can be made from the hardware or remotely using a WEB browser.

It is powered by a 12 VDC / 1.25 A switched power source and its internal battery provides up to 4 hours of autonomy.

1.1 LINK / TRANSMISSION

Link Types:

1. *Wired Ethernet.*
2. *Wi-Fi (using external antenna)*
3. *4G LTE (using USB modem)*
4. *Bluetooth (to a mobile phone)*
5. *Land line (POTS)*

Transmission methods:

1. *SIP calls (full duplex)*
2. *Streaming to icecast, shoutcast servers*
3. *Streaming RTP.*
4. *Phone call using smartphone (Bluetooth)*
5. *Phone call using POTS.*

1.2 RECORDING

MX2200 can record more than 100 hours of MP3 audio into its internal memory. The recordings are accessed by connecting the console to the LAN, and using a computer with a standard web browser.

1.3 AUDIO INPUTS

MICROPHONES

MX2200 allows to work with 3 dynamic microphones on 100 mm slide faders. The Fader #3 allows to use a fourth microphone connected to the input MIC-4. Note that MIC#4 do not has its own fader, so it will be useful only for certain situations.

An internal audio compressor on the microphone sub-mix minimize the volume variations to avoid overdrives and digital clipping.

LINE

MX2200 has two stereo LINE inputs: an analog line input (TRS) and the USB port for direct audio transfer from a computer.

1.4 TELEPHONE CONNECTIONS

MX2200 manages one conventional land-line and one cellphone linked via Bluetooth.

The telephonic communications can be used as the main link to the Studio or as a backup of the data link (streaming, SIP).

When a phone line is not being used as a link, it can be enabled for use in the program mix.

ATTENTION!
CONFERENCE IS NOT SUPPORTED

Conference calling between a landline and Bluetooth devices is not supported. While both callers may be on-the-air, they will not be able to hear with each other.

1.5 MONITORING and CUEING

Up to four headphones can be connected directly to the rear panel. **One knob set the level for all headphones**, so in order to set individual volumes, an external headphone amplifier should be used. A cheaper solution is to acquire headphones with built-in volume control.

The on-board talk-back microphone allows to speak privately to the phone lines or to all headphones.

The monitoring circuit (CUE) allows to the user to hear signals before mix them to the program, for example to dialogue with the phone lines.

Tally light

When a tally-light is connected to the console, the output HEADPHONES #1 behaves as a monitor output for loudspeakers, muting the signal when the microphones are on-air.

1.6 OUTPUTS

MX2200 has one stereo analog output, stereo PGM to USB and the CODEC that allows simultaneous streamings for SIP communication (full duplex) RPT and Icecast/Shoutcast server.

The main mix has a stereo (analog) insert point that allows to connect an external audio processor.

1.5 GENERAL WORKING DIAGRAM
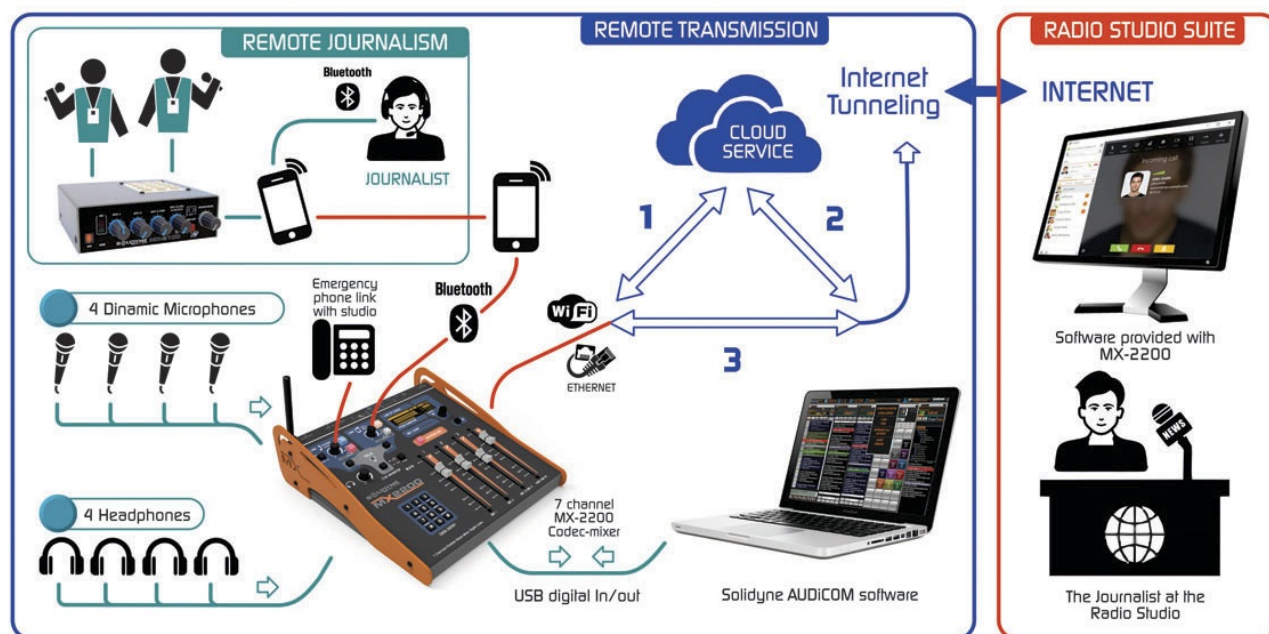
The previous diagram presents a working set that allows visualizing the flexibility of use.

The program signal can mix:

- Up to 4 dynamic microphones (two shares the fader MIC 3+4).

- 1 mobile phone: for a reporter who can directly use a smartphone with a hands-free set, or –in a more advanced implementation – use the mobile linked to a second console *Solidyne MX2100*.

- Audio from a laptop connected via USB. It allows to manage the whole contents from the remote location.

The link to the Studio:

- MX2200 gets Internet access:
  - connected to a Wi-Fi network.
  - connected to a wired LAN.
  - with a 4G MODEM.

- The main link to the Studio, in the example, is by a SIP call.

- SIP is full-duplex, with no audible delays.

- The user links the MX2200 to the Studio with a simple click: the console calls and the Studio takes the call (or vice versa). An Internet server (CLOUD SERVICE) connects both ends (1 and 2) and when the connection is established, the Server "leaves the scene" and the console stays in a direct-link with the Studios (3) through a " private tunnel" (Internet Tunneling).

- Simultaneously to SIP communication, the MX2200 CODEC can send two redundant streamings: one via RTP to a specific IP address and another one to a web server, with different audio qualities.

- The landline is used as backup link. The paired smartphone can also be used as a backup link when it is not used on the program mix.

Reception at the Studio:

- The SIP call is received using a standard computer, running a third-part software to manage VoIP calls. There are several free options (https://www.linphone.org/ is a great alternative).

Recording:

- MX2200 can records more than 100 hours of stereo program into its internal memory (MP3@192kbps). The recording can be made simultaneously with streaming.

2 Audio Operation

2.1 Overview

Audio sources, such as microphones, laptops, and phone lines, are connected to the console through the input channels. The mixer console merges all the signals to produce a single output signal, known as "Program" (PGM). The amplitude of each audio source are managed by the main faders. The final mix is transmitted to the studios via one or more transmission channels.

Different sections can be identified:

Figure 1: MX2200 areas

1 INTERCOM: This buttons allows to talk to the telephone lines, and to the headphones.

2 TELECOMMUNICATIONS: Allows to manage the land line and the Bluetooth (smartphone).

3 SCREEN AND COMMAND KNOB: Manages all internet-related features and settings, including streaming and SIP calls, as well as recording options.

4 KEYPAD: The keypad is used to enter characters in the configuration text fields. It also enables users to dial (DTMF) when using the landline.

5 FADERS: The main faders allows to mix the signals from microphones and line (USB or analog) to generate the program signal. The level should be maintained around 0 VU (-10 dBFS).

6 REAR PANEL CONNECTORS: These indications shows the locations of the main connectors on the rear panel, so that the connection is more user-friendly.

2.1.1 Power supply

The device is equipped with an internal battery that provides up to 4 hours of autonomy (refer to specifications). The battery level is displayed on the screen, and it is advisable to connect the charger once the battery level drops to 10%. The charging time for a full charge, when the battery is completely drained, is approximately 10 hours.

ADVICE
To charge the battery, MX2200 must be turned ON and connected to the power supply.

The equipment comes with a charger that provides 12VDC/2A and is compatible with 110/240V 50/60 Hz networks. It is safe to use the equipment while it is connected to the charger, even after the battery has reached 100% capacity.

In the event of a power outage, the mixer will automatically switch to battery mode.

2.1.2 ON/OFF

The MX2200 can be turned on or off using the retained push button located on the rear panel (refer to item 6 in Figure 1). Once the button is pushed to power on the device, a startup screen is displayed, which remains visible during system boot. After 15 seconds, the main screen appears.

To power off the device, simply release the button. The system will shut down after 5 seconds. It is recommended to wait a few seconds before turning the equipment back on after it has been turned off.

2.2 Display and encoder

The display and command knob are used to navigate through the system screens and options:

Turning the knob allows users to scroll through the screens and options within a screen.

A short press on the rotary control accesses the items on the current screen or allows users to edit a selected item. A short press on a value confirms a change.

Pressing and holding for 1 second cancels any changes made to an option or value.

The DATA ENTRY keypad is used to enter alphanumeric characters into the data fields as follows:

Successively pressing a key displays the characters for that key, in the currently selected position.

Turning the knob changes the position to be edited. Once all the characters have been entered:
- *A short press on the encoder confirms the input*
- *A long press cancels any changes made.*

For an overview, the four main screens are shown below and the sub-screens are listed; which will be detailed later.

1. MAIN SCREEN (HOME)

```
HOME            4G 品 令令爭
L                        -8 SIPRdy
R                        -9 RTP
Rx                       -90 Strm
```

PGM Level Indicators (L/R): Displays peak level in dBFS. The mark corresponds to 0 VU (-10 dBFS) and the indicator becomes brighter after that. It is essential to ensure that the signal does not permanently exceed this indication.

Receive level indicator (Rx): Shows the level of the audio return for SIP calls.

Status of 4G/Ethernet/Wi-Fi networks: The appearance of the icon indicates the correct connection to the local network.

Status of SIP, RTP and STREAMING connections: indicate when a transmission is active, off, or in an error condition (e.g., due to connectivity issue).

Battery level: It is recommended to plug the charger when the battery level drops to 10%.

Navigation: Pressing the Encoder at the HOME screen gives direct access to the SIP CONTACTS page.

2. CONNECTION CENTER

```
CONNECTION CENTER
SIP CALL Offline
RTP STREAM Dissabled
HTTP STREAM Dissabled
```
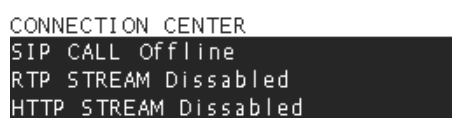
The display presents the current status of the three connection modes, allowing the user to establish or terminate each connection by accessing the corresponding option.

3. SETUP

```
SETUP
WiFi 0.0.0.0
Ethernet 192.168.0.127
Modem 0.0.0.0
```
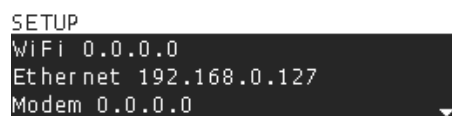
The SETUP screen shows the status and the IP addresses of the network connections. By pressing the encoder the user access to SETUP and is enabled to scroll all the settings options.

1. WI-FI: Shows available Wi-Fi networks.
2. ETHERNET: Wired connection configuration.
3. MODEM: Shows 4G MODEM configuration
4. RTP STREAM: Configuration for RTP streaming.
5. HTTP STREAM: Webcasting server configuration
6. SIP ACCOUNT: SIP account of the console.
7. SIP CONTACTS: SIP contact book
8. POWER INFO: Battery info
9. TIME ZONE: Time zone (for recording)
10. ABOUT: Firmware version.

The settings can also be accessed remotely via a WEB browser.

4. RECORDING

```
RECORDING
Status Stopped
Max time 30 minutes
Bitrate 256 Kbps
```
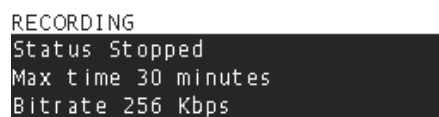
This screen shows the recording settings and enables the user to modify the values.

2.3 MAIN CHANNELS

Ilustración 2: Atenuadores principales

2.3.1 Microphones

The MX2200 have 3 microphone channels, plus the option of connecting a 4th microphone sharing the fader MIC 3+4.

The microphone gains are automatic, varying depending on the position of the fader. This automatic adjustment simplifies operation, maintaining an optimal signal-to-noise ratio and preventing preamplifiers from saturating.

The MASTER MIC button enables the microphone channels, which are sent to the PGM mix according to the fader level. It also activates the Tally Light and mutes the output of headphones #1 (when the Tally Light is connected).

The mics are not sent to the CUE mix. They are only heard in headphones when they are on the program mix (MASTER MIC on and fader raised).

2.3.1.1 Dynamic compressor

The dynamic COMPRESSOR starts to work when the microphone signal exceeds the threshold of -10 dBFS (0 VU). It operates with a high compression ratio, attenuating the signal up to 20 dB beyond the threshold.

The MIC COMP indicator displays the compression degree using a three-state signaling. The green indication begins to activate when the signal reaches the threshold (-10 dBFS). The red signaling indicates that the compressor is operating at its maximum attenuation, and the signal is nearing clipping (-5 dBFS). If the signal reaches the clipping condition, it will be displayed on the screen with a three-second hold.

When two microphones are connected to the dual MIC 3+4 channel, it is important for the signal to maintain the yellow indication and occasionally the red indication on the MIC COMP indicator.

ABOUT STEREO MIXES
While the MX2200 CODEC-MIXER has the capability to transmit in stereo, the mic channels generate a MONO mix.
To set up a stereo take on field, for example, for the ambient, an auxiliary audio console should be used, connected to the CODEC-MIXER MX2200 through the line input. In these cases, condenser microphones are commonly used, so the auxiliary console must have a phantom power.

2.3.2 USB audio and analog line input

When a computer is connected via USB, the USB/LINE IN fader is used to manage the USB signal. When the USB is disconnected, the USB/LINE IN fader receives the LINE IN stereo analog input.

In the former case, the LINE IN input is accessible through the phone line knob (PHONE), which allows to toggle between PHONE and LINE IN. Thus, the user can configure a setup with microphones, two stereo lines, and a mobile phone (Bluetooth).

To send the USB/LINE IN signal to the PROGRAM mix, just raise the fader. To cue the signal on USB/LINE IN (with fader closed) press the button CUE located over the fader.

AUDIO USB
When connecting the USB port to a computer, the operative system will install a USB recording device, which receives the PGM mix from the mixer.

2.4 MONITORING

Image 3: Monitoring

The knob located on the CUE MONITOR area set the **volume for all headphone outputs**.

The "**PGM**" **button**, also located in this area, mutes the program signal on the headphones, allowing the user to listen to the signals on CUE. The signal CUE is always routed to all headphone outputs.

To work with individualized volume for each headphone, **an external headphone splitter** must be used, connected to any of the headphone outputs. Alternatively, headphones with built-in volume control offer a more cost-effective solution.

2.4.1 Output for active loudspeakers

When a load is connected to the Tally-Light output, the "Headphone #1" output becomes a Monitor Speaker Output. This output is muted when the microphones are enabled to avoid acoustic feedback's with the loudspeakers. Headphones outputs #2; #3 and #4 remain unaffected.

2.4.2 Talk to CUE

By pressing the TALK button located in the CUE MONITOR area, the audio signal in the headphones is muted, and the signal from the console's built-in microphone is heard. This feature enables the console operator to communicate with other team members while wearing their headsets, without the need to wave, raise their voice, or take off their headsets to have a conversation.

2.5 TELEPHONE LINES

2.5.1 Overview

MX2200 manages one conventional landline (POTS) and one mobile phone, linked via Bluetooth. Both can be used as:

- Back up link to the Studios, when MX2200 is streaming via Internet.
- The main link to Studios when Internet is not available.
- An audio source to add a call to the PGM mix.

2.5.2 USING THE PHONE LINES FOR "ON-AIR" CALLS

- The lines can be used to add another participant, such as a field reporter, or an interviewee; to the broadcast.

- A calling can be aired using either the fixed line or the Bluetooth mobile, while the console is linked to the Studio via Internet. When a phone line is being used as audio link (backup or primary) the available one can be used for on-air callings.

- While is on-the-air (in PGM) the phone line will be able to interact with the console operator and with people at the Studios.

- The return from the studio is mixed to the CUE and sent to landline and Bluetooth.

- The audio level for the return from Studios can be settled on SETUP options.

2.5.3 THE PHONE LINES AS STUDIO LINK

- Once the calling is made, the Master Studio receives the PGM signal through the telephone line.

- If the PGM/CUE button is set to "CUE", the signal that returns from the Studio is heard on CUE monitoring. In this condition, talkback microphone will be sent to the Studio pushing the TALK button of this line.

- When the PGM/CUE button is set to "PGM", the return from the Studio is suppressed from CUE and sent to the PGM mix.

- The signal sent to the phone line (the same is for Bluetooth) is mix-minus, meaning that the return signal from the Master Studio is not mixed to the PGM signal that is sent to the Studio. This is done in order to prevent a feedback loop.

- When the LINE is set to PGM, the mic "TALK "is not sent to the Studio.

- To avoid the talk-back microphone from accidentally going on-air while the phone line is functioning as a studio link, ensure that the PGM/CUE button is set to "PGM."

- To listen to the Master Studio the user can increase the volume of the LINE to mix it with PGM. The return signal from the Master Studio can be heard through the headphones, and it will not cause feedback since the send is mix-minus.

- If you do not need to hear the return from the studio, keep the PGM knob closed.

2.5.4 CONTROLS (landline and Bluetooth)

The following procedures are valid both for adding calls to the local transmission and for using the lines to link to the Studio.
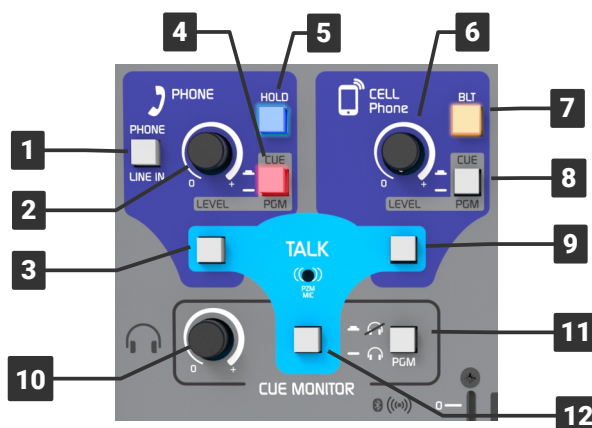
Figure 4: Hybrid's and Bluetooth link

1 The PHONE/LINE IN button switches between the telephone line and the stereo LINE IN.

When the MX2200 is linked to the Main Studio via the Internet, the two microphone channels and two audio sources (LINE IN and USB) can be mixed.

When PHONE/LINE is set to "LINE IN," incoming calls will still be indicated by the flashing LED, enabling the operator to switch to PHONE mode to answer the call (refer to the later instructions for more details).

2 The knob LEVEL adjusts the volume for either the phone line or the LINE INput. The level will be for CUE or PGM depending on the position of the PGM/CUE button (see POINT 4).

3 Push button TALK allows to talk to the land line, using the built-in microphone.

4 CUE/PGM switches the signal (PHONE line or LINE IN) between the headphones mix (CUE) and the program mix (PGM).

5 The "HOLD" button serves to indicate incoming landline calls with a flashing ring and enables users to answer calls by taking the line. If the CUE/PGM button is set to "CUE," the caller will be placed on hold and the audio will be heard on headphones. If the CUE/PGM button is set to "PGM," the calling will be mixed to PGM output. In both cases the knob LEVEL set the mix level.

6 This knob sets the audio level of the Bluetooth audio signal, which is mixed to either the CUE or PGM out depending on the status of the CUE/PGM button.

7 Turn on/off the Bluetooth device on the MX2200. It also allows to activate the "pairing" mode to link a mobile phone (see below).

8 Switches the Bluetooth's audio signal between the CUE listening and the program output (PGM).

9 TALK is a push-button that allows to talk to the linked cell phone using the MX2200's built-in mic.

10 This knob set the level for all headphones.

11 The PGM switch located in the CUE MONITOR area mutes the program audio signal in the headphones, allowing for exclusive listening to CUE signals. This feature is particularly useful, for example, for engaging in a phone conversation or monitoring only the return audio from studios during a SIP call.

12 Is a push-button that allows to talk to the headphones, using the built-in microphone.

2.5.5 USING A BLUETOOTH CELL PHONE

The MX-2200 can establish a Bluetooth connection with any compatible mobile phone.

There are three actions that should not be confused when working with Bluetooth:

ENABLE/DISABLE BLUETOOTH ON THE CELL PHONE

It refers to whether Bluetooth is turned on or off on the phone.

PAIR THE CELL PHONE WITH A BLUETOOTH DEVICE

It refers to the process where the mobile device finds and registers an external Bluetooth device (for example, headphones, the car audio system,

or a BLT200 channel). This procedure is performed only once for each device. Once paired, the device can be unpaired, which means the phone "forgets" it, and the pairing process must be repeated to connect it again.

CONNECT/DISCONNECT A BLUETOOTH DEVICE TO THE CELL PHONE:

It refers to the action where the phone connects to a previously paired device. The action of connecting/disconnecting should not be confused with the action of pairing/unpairing.

When the mobile phone has Bluetooth enabled, it can easily connect and disconnect from different Bluetooth devices by simply selecting the device from the list of previously paired devices (if there are Bluetooth devices present that have not been paired, they will not appear in this list).

2.5.5.1 PAIRING THE MOBILE PHONE

To pair a Bluetooth channel with a mobile phone, a very similar procedure is followed to that used for pairing other Bluetooth devices, such as speakers or hands-free headsets.

The BLT button are illuminated and flash at different rates depending on the Bluetooth status.

BLUETOOTH BUTTON FLASHING MODES		
A	CELL PHONE CONNECTED	Short flashing (1s) in yellow with long off (2s) = Indicates it is connected to the cellphone without an ongoing call at that moment.
B	CELL PHONE DISCONNECTED ACTIVE CALLING	Slow flashing (2s) in yellow with short off (1s) = indicates an ongoing call. Upon ending the call, it returns to mode "A". It also indicates Bluetooth is on without a mobile phone connected. When a phone is connected, it switches to mode "A".
C	PAIRING MODE	Fast flashing in yellow. This mode allows a phone to find and pair with the BLT200.
D	INCOMING CALL	Continuous blue light.

To pair a cell phone, proceed as follows:

1. **On MX2200:** Verify that the Bluetooth is turned off (BLT button not illuminated). When Bluetooth is on, the button flashes slowly intermittently (modes A or B). If it is on, press and hold the button until the LED turns off completely.

2. **On MX2200:** Activate pairing mode (this mode can only be accessed starting from Bluetooth off). Press and hold the BLT button (approximately 8 seconds). After a few seconds, you will see the yellow light flashing slowly (mode B), and later the flashing becomes fast (mode C). Release the button only then. The continu-

ous flashing indicates that pairing mode has been activated, and it is ready to be found by the mobile phone.

3. **On cell phone:** Turn on Bluetooth and perform a search for new devices. The procedure varies depending on the make and model of the phone (if in doubt, refer to the phone's documentation).

4. The MX2200 Bluetooth device will appears as **Solidyne BTM641**.

5. Select Solidyne BTM641 from the list to connect it.

<div style="background:red;color:white;text-align:center;font-weight:bold">BLUETOOTH DEVICE NAMING</div>

The user can edit the default Bluetooth name "Solidyne BTM641" on the mobile phone, usually from an option in the Bluetooth settings (refers to the cell phone user's manual)

Once the devices are paired, the BLT button keeps flashing slowly (Mode A) indicating that Bluetooth is connected.

It is not necessary to repeat this procedure while using the same phone. To pair a new phone, repeat the whole procedure.

<div style="background:orange;color:white;text-align:center;font-weight:bold">IMPORTANT</div>

Multiple mobile phones can be paired with the MX2200's Bluetooth, but two mobile phones cannot be connected simultaneously to the unit.

<div style="background:red;color:white;text-align:center;font-weight:bold">DISTANCE TO CELL PHONE</div>

While the mobile phone can be up to 3 meters away, it is recommended to keep it nearby, but never directly on the BLT200 interface.

2.5.5.2 RE-CONNECT A PAIRED MOBILE PHONE

Just like with any conventional Bluetooth device, the mobile phone will "disconnect" from the BLT200 in the following cases:

• Disabling Bluetooth on the phone. Typically, this action does not end an ongoing call; it transfers it to the phone. However, this may vary depending on each brand and model. Reconnection is restored when Bluetooth is turned on on the phone. In some cases, reconnection is automatic. In others, it will be necessary to choose the SOLIDYNE device again from a list. If the call was not interrupted, upon reconnection, it will return to codec/mixer.

• When a call is in progress and the phone disconnects from the MX2200's Bluetooth device, the call is transferred to the phone. If it reconnects to Bluetooth, the call returns to the codec/mixer.

• When during an ongoing call, the "Speakerphone" mode is activated, this action also removes the call from MX2200 and transfers it to the phone. If Bluetooth is reconnected, the call returns to codec/mixer.

• If the Bluetooth on MX2200 is turned off.

In any case, re-connection will not be possible if the same Bluetooth channel on BLT200 was paired with another phone and it is now connected.

2.5.5.3 VOLUME ADJUSTMENT

To achieve optimal reception and the highest signal-to-noise ratio, it is essential to set the Bluetooth volume on the mobile phone to the maximum level. This step is critical, as it ensures that the signal is strong and free of noise.

<div style="background:red;color:white;text-align:center;font-weight:bold">BLUETOOTH AUDIO LEVEL</div>

The volume for Bluetooth must be set with Bluetooth enabled and connected to MX2200. The volume of the mobile phone's speaker is independent of the Bluetooth device. If you adjust the volume without making a Bluetooth-enabled call, it only changes the speakerphone volume, not the Bluetooth volume.

The Bluetooth volume is stored by the mobile phone. If you change the phone, you must repeat the operation to set the Bluetooth level to the maximum.

2.5.5.4 TURN OFF MX2200's BLUETOOTH

To turn off the Bluetooth on the mixer-codec, press and hold the BLT button until the flashing stops.

2.5.6 MIX BLUETOOTH TO PROGRAM

1. **On the console:** On the sector "CELL Phone", the knob LEVEL must be closed and the CUE/PGM button must be set to CUE.

2. **On the console:** Turn on Bluetooth by pressing and holding BLT until the button lights up. The button keeps flashing slowly indicating that Bluetooth is active.

3. **Enables Bluetooth on the mobile phone.** If the phone was paired with the console, the link is restored and in a few seconds the mobile phone is linked to the console.

<div style="background:red;color:white;text-align:center;font-weight:bold">RECONNECTION</div>

In some cases the re-connection with the Bluetooth devices is not automatic, and the user needs to choose the device in the mobile phone to "connect" it.

In case that no device connects, MX2200's Bluetooth will automatically turn off after five minutes.

2.5.6.1 INCOMING CALLS

1. To prevent accidental broadcasting when answering a call, it is essential to set the CUE/PGM button to the CUE position (button released). This step is critical because if the button is set to the PGM position, it may cause the call to be aired.

2. When the mobile receives a call: It can be answered directly from the console with a short touch on the BLT button.

3. Once answered, use the TALK button and the headphones to dialogue with the caller.

4. To mix the call to the program output, set the button CUE/PGM to PGM and adjust the level.

5. Communication can be ended with short touch on BLT button; or from the cell phone.

2.5.6.1 DISCONNECT/RE-CONNECT BLUETOOTH

When working with Bluetooth, there are three actions that should not be confused:

Enable/disable Bluetooth on the mobile phone:

Refers to the process of activating or deactivating on/off the Bluetooth function on a mobile phone.

Pairing the mobile with a Bluetooth device:

Refers to the process in witch the mobile searches for an external external Bluetooth device (such as headphones, a car's audio system) to storage its unique ID code. This procedure is only necessary once for each device. Once paired, the device can be unpaired, which essentially erases its stored ID code from the mobile phone. To reconnect it, the pairing procedure must be done again.

Connect/disconnect a device to the mobile:

Refers to the process of establishing or terminating a connection between a mobile and a previously paired Bluetooth device. This process should not be confused with pairing/unpairing, which is the initial process of creating a link between two Bluetooth devices. When Bluetooth is enabled on the mobile, it can easily connect and disconnect from different Bluetooth devices by simply selecting the device from the list of paired devices (if there are present Bluetooth devices that were not paired, they are not shown in this list).

Just like with any other standard Bluetooth device, the mobile will disconnect from the console in the following situations:

- **By disabling Bluetooth on the cell phone.** Disabling Bluetooth on the mobile phone typically does not terminate an ongoing call, but instead transfers it to the phone, although this behavior may differ across models. To re-establish the connection, Bluetooth can be enabled on the phone. Re-connection is automatic in some cases, while in others, the user must manually select the SOLIDYNE BTM641

device from a list. If the call was not interrupted, it will be accessible on the console again upon re-connection

- During an ongoing call, when **the phone is disconnected** from the Bluetooth device (the console). The call is transferred to the phone. If Bluetooth is reconnected, the call returns to the console.

- During an ongoing call, **when the "Speaker" mode is activated**. This action also takes the call off the console and transfers it to the phone. If Bluetooth is reconnected, the call returns to the console.

In any case, re-connection will not be possible if the Bluetooth on the console was paired with another phone and it is now connected.

The next image shows a screen capture of an Android mobile with a call. Enabling the option "Headset" the call transfers to the Bluetooth device (usually a handsfree headset); or returns it to the mobile when it is disabled (refers to phone's info).
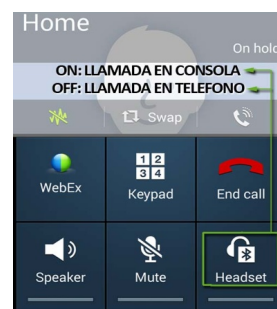
Figure 5: Bluetooth modes on calling

2.5.6.2 GENERATE THE CALL ON THE MOBILE

To generate the call from the mobile phone:

1. *On the mobile, change the Bluetooth mode to "Speaker", to talk from the mobile.*

2. Make the call (SEND).

3. To transfer the call to the console, return to "Headset" (Bluetooth) mode on the mobile.

4. Verify before transferring the call that the CUE/PGM button is in CUE mode (released). If the button is in PGM, when transferring the call it could accidentally go on air.

5. To send the mobile phone on the air, change CUE/PGM to PGM and adjust the level.

6. To end the communication, briefly press the BLT button on the console (it can also be ended from the mobile).

2.5.7 LAND LINE (POTS)

2.5.7.1 RECEIVE THE CALL

1. Upon receiving an incoming call, the HOLD button will flash in sync with the ring cadence, regardless of the position of the PHONE/LINE IN button.

2. Prior to answering the call, ensure that the PGM/CUE button is in the CUE position.

ATTENTION!
If the PGM/CUE button is assigned to PGM when answering the call, the call will be mixed directly to the PROGRAM output with the level set by the LEVEL knob. Therefore, it is essential to verify that the PGM/CUE button is in the desired position before answering the call.

3. In order to hear audio from the phone line and for the line to receive return, the PHONE/LINE-IN button must be in the PHONE position (released).

4. To answer the call, press the HOLD button. The caller will hear the PGM signal return.

5. The level in the headphones can be adjusted with the LEVEL knob, and the CUE MONITOR level setting is also involved.

6. To speak to the person on the other end of the line, press the corresponding mic TALK button. By pressing TALK, the caller will stop listening to the program mix, and the operator can talk with the caller by holding down the button. When released, the line remains on hold with program signal.

7. The PGM button in the CUE MONITOR section mutes the program signal from the headphones, enabling to the user to hear only the signals from the telephone line.

8. To send the call to the PROGRAM, set the PGM/CUE button to PGM and adjust the mixing level with the LEVEL knob. The caller will be then able to talk with the MX2200.

9. To take the line off the air and put it on hold, turn the LEVEL knob to zero to mute it, then press CUE/PGM to switch to CUE. The line will be retained, listening to the signal from the program.

10. To end the call and release the line press HOLD.

2.5.7.2 START A CALL FROM THE CONSOLE

1. Verify that the PGM/CUE button is in the CUE position (released).

2. Take the line by pressing HOLD. The dial tone will be heard in the headphones, and the volume of the tone can be regulated with the LEVEL knob located on the PHONE LINE area. It should be noted tha the CUE MONITOR level knob is also involved.

3. Dial the number using the console keyboard. If a hang-up is required to redial, release the HOLD button, and press it again to pick up the line again.

4. Once the call is established, continue from point 6 detailed above.

2.6 Recordings

The MX2200 can record the PGM signal. Recordings are saved in an internal memory as stereo MP3 files (16 bits / 48 KHz). The RECORDING screen enables the user to START/STOP recording and to change the recording settings.

```
RECORDING
Status Stopped
Max time 120 minutes
Bitrate 192 Kbps
```

The **STATUS** option allows the user to ENABLE or STOP the recording. When a recording in progress is stopped, the audio file is saved up to that point.

```
Enable recording
   Yes              No
```

The **MAX TIME** option allows the user to define a maximum recording time. This way, the operator does not need to remember to stop the recording. The recording will end automatically when this time is reached.

```
Max time
Use keypad to edit
              120
```

To edit this value, press the numeric keypad on the console to enter the desired value, up to a maximum of 240 minutes (4 hours). Press the knob to confirm the new value. It is also possible to increase the recording time limit while a recording is in progress.

BIT RATE set the audio quality for the recordings, with higher values indicating better quality but larger file sizes.

```
RECORDING Quality
256 Kbps
192 Kbps
160 Kbps
```

The available range is between 64 Kbps and 320 Kbps. At 192 Kbps, the MX2200 can store approximately 100 hours of audio.

When the recording is enabled, the remaining time is displayed on the main screen of the MX2200 console. This allows the operator to keep track of how much recording time is left before the recording stops automatically or before the user manually stops it.

2.6.1 Accessing the recordings

The recorded audio files can be accessed using an external computer and a web browser (such as Firefox, Chrome, Safari, Edge, etc.). In address bar enter the device's IP address followed by "/recording". For example:

<div align="center">

192.168.1.10/recording

</div>

The user can access the recording management interface and perform various tasks such as playback, renaming, downloading and cleaning recordings remotely.

WEB ACCESS
1. Connect the MX2200 console to the local network via the Ethernet port (recommended) or Wi-Fi antenna.
2. Open a web browser on a computer connected to the same network and enter the IP address of the MX2200 to access its web interface.
3. To find out the IP address, turn the CONTROL knob (encoder) to display the SETUP screen. The first three items show the connection status and the assigned IP address for Ethernet, Wi-Fi, and Modem.

The file name of the recordings will include the *host name, date, and time*. The web interface (HTML5) includes an audio player.
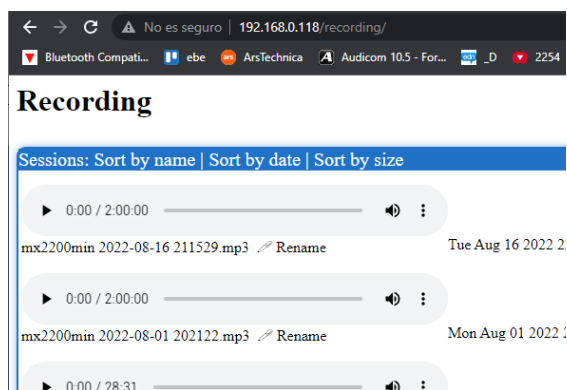
The list can be sorted by name, date, and size. The recording date is displayed in the local time of the computer on which it is being viewed. **The audio files can be downloaded** from the right-clic menu (to the browsers default folder).

3 Link with the Studios

3.1 INTRODUCTION

The link to the studios can be established in several ways:

1. Sending digital audio via the Internet. It is accessed from the control screen (Connection Center) and there are three options:

   a) VoIP call (two-way) with minimal delay using a SIP account (provided by third parties)

   b) Direct RTP streaming to the studios IP address (streaming only).

   c) Streaming to a Shoutcast/Icecast server (streaming only).

2. Phone call using a mobile phone linked by Bluetooth.

3. Telephone call using land line.

"In the first scenario, once the Internet link options have been configured, the field user can easily establish a connection with the MX2200 by simply connecting it to the Internet and initiating the corresponding link. No additional configuration or advanced technical knowledge is required.

Furthermore, the three available connection methods can be combined to provide additional functionality, such as establishing a backup link; for example, using a SIP call as the main link and a cell phone call as a backup. Or complementing the main link, for example, an RTP streaming as main link and a telephone call as auxiliary link, to communicate with the producers at the Studios"

3.2 Internet connection

The MX2200 CODEC-MIXER can connects to the Internet through several different methods, but **only one connection can be active at time**. Although the simultaneous use of several connections (bounding) is not supported yet; it is possible to configure multiple internet access methods, witch enables automatic failover to another method in case of a connection failure.

The ways to connect MX2200 to the Internet are the following:

3.2.1 Connecting to a Wi-Fi network

1. Connect the factory-supplied Wi-Fi antenna to one of the USB ports labeled "Wi-Fi".

   If the MX2200 recognizes a Wi-Fi network previously used, it will automatically connect as long as the credentials are still valid.

Figure 6: WiFi antenna. The model supplied with the unit may differ from the image.

2. Ensure that the Wi-Fi icon displayed in the upper section of the main screen confirms that the Wi-Fi antenna has been recognized. The icon will remain greyed out until a successful Wi-Fi connection has been established."

3. Rotate the encoder to access the SETUP screen and press to access the available options.

4. Press on the Wi-Fi option to display a list of available networks.

5. Select the desired network and press to establish the connection.

6. A screen to enter the network password will be displayed.

   To enter the password, use the DATA ENTRY keyboard (see details at the beginning of this section).

7. After the password are validated, the network will assign an IP address to the unit, witch will be displayed next to the network name under SETUP/Wi-Fi. If the password is invalid, the IP address will remain at zero. In this case, the procedure should be repeated from Step 4.

8. Remember: to cancel and exit any screen, press and hold for 1 second.

9. The MX2200 is now connected to the Wi-Fi network. The Wi-Fi icon on the home screen will display the signal strength of the network.

10. The unit is now ready to start transmitting data via SIP call, RTP streaming, or connection to a web server, depending on the specific configuration and requirements.

11. Note1: The user can verify if the LAN has internet access by connecting a mobile phone or notebook to the network and browsing the internet.

12. Note2: It is important to note that a mobile phone can be used as a hotspot for internet sharing. This allows the mobile phone to generate a Wi-Fi node to which the MX2200 can connect.S

> **ADVISE!**
> **OPEN PUBLIC NETWORKS**
>
> Always verify the actual connectivity of the network to which the unit is connected. In the case of open public Wi-Fi networks with web logins, the unit may falsely report an OK Internet connection without properly authenticating.

3.2.2 Connecting to a wired network

1. Connect the MX2200's Ethernet port to a socket on the local network using a standard UTP cable.

2. By default DHCP is enabled, so the network router/switch will automatically assign an IP address to MX2200.

3. Verify on the SETUP screen that the unit has an assigned IP (other than 0.0.0.0).

4. Note: The user can verify the Internet access of the network connecting a notebook with Ethernet port and then browsing the Internet.

3.2.3 Connecting to a 4G/5G network

1. The unit requires an external 4G/5G USB MODEM (not provided).

2. Connect the MODEM to one of the ports indicated in the image, using a USB extension cable with a minimum length of 30 cm.
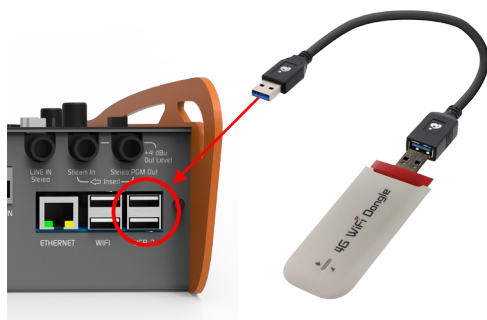
WARNING! The direct connection of the MODEM to the console can cause interference in the audio.

3. Once the MODEM is detected, the 4G icon is displayed on the main screen.

4. The UNIT is ready to start the transmission by SIP call, RTP streaming or connection to a WEB server, depending on the case.

3.3 SIP CALLS

This option provides real-time full-duplex communication over the Internet, similar to a conventional VoIP call such as Skype or WhatsApp, but with full control on the audio quality.

It the best option for situations where real-time interaction with the studios is necessary and requires simplicity in the Internet connection, without any need for network configuration at either end. This mode is commonly used for transmitting from locations where there is no control or security over the conditions of the available Internet access, such as bandwidth, stability, or access restrictions.

The resulting communication is a streaming audio that is sent directly from the console to the Studio, and can be configured to use low bit rates when Internet bandwidth is poor. When set to maximum quality, the audio quality is equivalent to the sound of the microphones on the Studios.

Here's a revised version of the text in a more technical English style:

SIP calls on the MX2200 are extremely user-friendly. A pre-loaded phonebook displays available contacts to the user. To establish a link with the Studios, the user selects a contact and initiates the call. At the Studios, an operator receives the call on a computer. From the Studios, the operator can also call the MX2200. Technical details for this process are available in Section 4 of this user manual."

3.3.1 Making a SIP call

1. Using the Encoder, go to CONNECTION CENTER.

2. Press the Encoder to access the options.

3. Select the option SIP CALL and press. The SIP contacts book will be displayed. This screen can also be accessed by pressing Encoder on the HOME main.

4. **Rotate the Encoder until select the desired contact.**

5. Press on the contact to start the calling.

6. When the operator at the Studios answers the call, the screen will indicate SIP CALL ENABLED.

7. To end the call, go back to CONNECTION CENTER and press again on SIP CALL.

3.3.2 Receiving a SIP call on MX2200

To receive a SIP call on the MX2200, the operator at the studios calls to the remote MX2200. When the MX2200 console receives a call, a dialog box pop ups giving the option to answer or reject the calling. At the same time, a ring is played in CUE to alert the user of the incoming call.

3.3.3 Audio return from the Studios

The MX2200 console sends the program mix to the Main Studio.

The Main Studio audio (also known as *studio return*) is directly added to the CUE circuit.

The signal sent by the Main Studio is mix-minus, meaning it does not contain the MX2200's mix.

The MX2200 internally mixes the PGM signal and the studio return signal on CUE. The user has the option to remove the PGM signal from the CUE mix to hear only the studio return. Refer to section 2.3, Monitoring, item 11 for further details.

3.3.4 Reception of MX2200 in the Studios

The SIP call is received on a computer that is running SIP-compatible VoIP software. There are various free tools available that can be used for this purpose. We recommend using LINPHONE, which can be downloaded from www.linphone.org.

In the VoIP software, you will need to configure the SIP account that was created for the Studio. This contact is the one that the console will call. Additionally, you will need to configure the playback and audio input devices so that they are linked to the studio's console.

For more details on the setup process, please refer to the Technical Notes section.

3.4 Streaming RTP

An RTP streaming connection involves greater network configuration complexity for both reception and transmission on the console's connected network. If the connection conditions are optimal, it is straightforward for the user. However, if issues arise, network configuration expertise is required to troubleshoot and resolve them.

RTP streaming is typically used in situations where the transmission originates from fixed installations with known configurations and requires the highest audio quality.

To establish the RTP connection (once the network has been configured and the IP accesses have been obtained) proceed as follow.

8. Using the encoder go to the CONNECTION CENTER.

```
CONNECTION CENTER
SIP CALL Offline
RTP STREAM Dissabled
HTTP STREAM Dissabled
```

9. Press the Encoder to navigate the options. Select RTP STREAM and press.

```
CONNECTION CENTER
SIP CALL Offline
RTP STREAM Dissabled
HTTP STREAM Dissabled
```

10. A screen will appear that allows to enable/disable the RTP link.

```
RTP Enable

    Yes              No
```

11. Pressing YES on the RTP link screen will establish the connection with the Studios. Once the connection is established, both the CONNECTION CENTER and HOME screens will indicate RTP STREAM ENABLED.

12. To end the RTP link, repeat the above steps and select NO (disable).

3.4.1 Return from the Studio on the console

The RTP broadcast is unidirectional, meaning that the transmission is one way only. Generally, in RTP transmissions, there is no need for a return signal from the Studio since the signal is generated from the location and there is no interaction with the Studio.

If you wish to receive returns from the Studios, you must establish a telephone communication (mobile or fixed line) or a SIP call.

3.4.2 Reception of MX2200 in the Studios

In order to receive streaming from an MX2200 console using a computer, software capable of receiving streaming with support for the RTP protocol is required. We recommend using VLC Player, which is a free media player (http://www.videolan.org/vlc/).

The audio format must be set to PCM16@44.1Khz or one of the MP3 modes. These formats allow for direct playback in VLC Player. However, if the OPUS CODEC is used, a load file (payload) of the type .SDP that describes the content is required.

To begin playback in VLC Player, navigate to the Media menu and select "Open network location". In the URL field, enter the following information:

[protocol]://[source_IP]:[port]

For example,

RTP://0.0.0.0:3030,

where RTP is the protocol used by the MX2200 for transmission, '0.0.0.0' indicates that the stream can come from any incoming IP, and '3030' is the internet port used.

Please note that some versions of VLC Player also support the RTP://@:3030 command.

3.5 Stream to a media server

A media server is a cloud-based service provided by third-party vendors and is usually contracted based on the number of simultaneous users who can connect to the server (we recommend consulting specialized technicians).
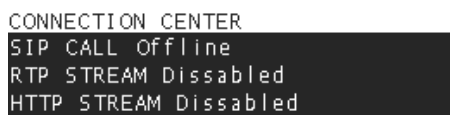
The media server receives the audio broadcast from the MX2200 and re-sends it through a public link that can be accessed from anywhere in the world by multiple end-users simultaneously.

Similar to SIP calls, its main advantage is the simplicity of connection for the user. Once the media server is configured in the MX2200, it does not require others configurations in the remote location's network. The MX2200 only needs to connect to Internet and start the "HTTP STREAM" broadcast.

This streaming method has considerable delays, between 5 and 30 seconds depending on the case. Therefore, it is only used when the signal is generated from the remote location without interaction with the Studios.

To establish the RTP connection:

1. Go to the CONNECTION CENTER screen, turning the Encoder.

```
CONNECTION CENTER
SIP CALL  Offline
RTP STREAM Dissabled
HTTP STREAM Dissabled
```

2. Press to navigate the options. Select HTTP STREAM and press.

```
CONNECTION CENTER
SIP CALL  Offline
RTP STREAM Dissabled
HTTP STREAM Dissabled
```

3. A screen will be displayed that allows you to enable/disable the link.

```
HTTP Stream Enable
     Yes              No
```

4. Press YES to start the link with the WEB SERVER. Once the connection is established, the CONNECTION CENTER and HOME screens will indicate HTTP STREAM ENABLED.

5. To end the link, repeat the above steps and select NO (disable).

3.5.1 Reception of MX2200 in the Studios

The transmission from a web server is the easiest to receive, as it is an http link provided by the provider, which is downloaded using any web browser.

3.6 Telephone call

Landline phone connections (also known as POTS) and using a mobile phone are now standard on all streaming consoles. In MX2200 they are done in a simple way. See "2.5 TELEPHONE LINES" in the operation section.

In the Studios the call is received using conventional telephone hybrids.

3.6.1 Using the lines as a link to the studio

When the equipment broadcasts over the Internet, the land line or a mobile phone can be used as a backup transmission method.

The lines can also be used as the main transmission channel, but the audio quality will be restricted to the telephone line.

Considerations for the use of the lines as links to the Studio:

• Once the call is established, the Studios always receive the PGM mix, through the telephone line. It does not matter if the line is in

CUE or PGM mode (see 2.5.4 TELEPHONE LINES/CONTROLS).

LINE AS BACKUP

- When the line is in the CUE position the studio return is sent only to the CUE monitor mix. If the TALK button is pressed on that line, the command microphone is sent to the line (to the studios).

- The above condition allows the MX2200 user to chat with the Studio while the console broadcasts over the Internet.

LINE AS MAIN LINK

- When the line is in PGM position, the Studio return is suppressed from CUE monitoring and added to PGM.

- The TALK mic is not sent to the Studio.

- **When a line works as the main link, it should be used in the PGM position to prevent the command microphone from accidentally going on the air.**

- If you don't need to hear the Studio return, keep the PGM knob closed.

- To hear studio feedback, turn the PGM knob level. The studio signal is heard in all headphones monitoring the PGM mix.

- The MX2200 sending to the lines is mix-minus, that is, the signal coming from the Studio is suppressed from the PGM mix that is sent to the Studio, to avoid a feedback loop.

3.6.2 Reception in the Studios

Calls are received at the Studio using conventional telephone hybrids.

All of the Solidyne line of consoles have the ability to connect to standard telephone lines and Bluetooth interfaces to link mobile phones.

4 Setup and connections
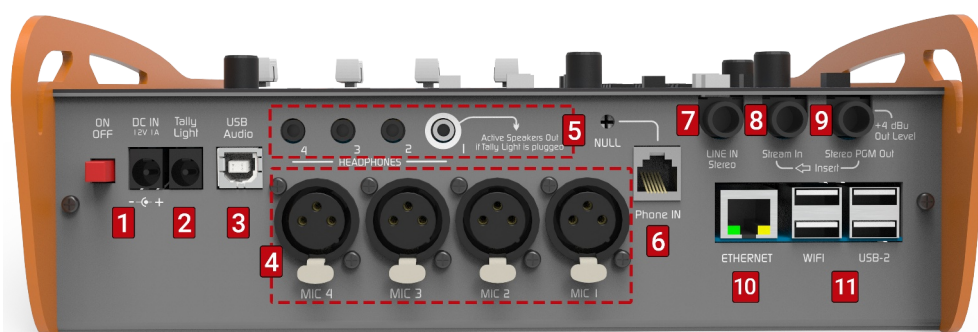
4.1 GENERAL DIAGRAM

4.2 REAR PANEL

Ilustración 7: Vista trasera de MX-2200

4.2.1 POWER SUPPLY

1 The unit includes the switching power source.

Input: a 90/240V 50/60 Hz
Output: 12 V / 1.5 A output.

The input **DC IN** uses a 2 mm tubular connector (center pin +).

⚠️ THE USE OF EXTERNAL POWER SOURCES OTHER THAN THE ONE SUPPLIED FROM THE FACTORY WILL CAUSE THE LOSS OF THE WARRANTY.

The unit is equipped with an internal battery that provides approximately 4 hours of autonomy (this value was estimated based on a SIP call active via Ethernet, with no Wi-Fi dongle and Bluetooth turned off).

To charge the battery the MX2200 must turned on. The full charging time, if the battery is completely discharged, is approximately 10 hours.

The unit can also be powered by connecting an external 12 VDC battery to DC IN. Take special care to respect the polarity.

4.2.2 TALLY LIGHT

2 The Tally Light output supplies a 12 VDC/200 mA power source for connecting an ON-AIR signaling light. It uses a tubular connector with a center pin of 2.5mm and a positive polarity.

When the console detects an air light connection, headphone output #1 works as a monitor speaker output, muting each time the air light is activated.

4.2.3 USB AUDIO

3 The USB port is to connect a computer. Use a USB type A/B cable.

Please use the cable supplied with the unit. If necessary, longer or extension patch cords can be used; but we recommend not exceeding 4 meters. Additionally, it is very important to use good quality cables.

To connect the MX2200 to a computer via USB, please follow these steps:

1. Connect the power supply to the console.

2. Connect the USB cable to the console.

3. Connect the USB cable to a USB port of the computer.

⏚ Make sure that the computer is properly grounded, to avoid potential damages on the USB ports.

MX2200 USB device will be automatically installed on Windows© 7/10, Linux Ubuntu and Apple OS systems. The operating system will detect a new USB connection and install two devices:

- "USB playback device" is the USB audio input from the side of the console

- "USB recording device" is the USB audio output on the console. This recording device receives the PGM signal.

Remember: In Windows, the playback and recording devices can be visualized in *"Control Panel → Sound and Audio Devices" → "Audio"*. There the user can define the default playback and recording devices (used by Windows applications).

4.2.4 MICROPHONES

4 The MX2200 features three microphone channels, each with balanced input for dynamic microphones located on the rear panel.

The input MIC#4 shares the preamp and the fader with MIC#3. To achieve signals with similar level (assuming similar sources such us environmental sound or two peoples with similar voices) we recommend using two microphones of the same make and model connected to the paired inputs MIC-3/MIC-4.

4.2.5 HEADPHONES

5 The MX2200 has four headphone outputs, each with a TRS (1/8") minijack connector. All outputs send the same level, which can be controlled with the knob located in the CUE MONITOR zone.

Headphone output #1 can also be used to connect monitor speakers, and when an air light is plugged in, it is detected by the console. Whenever the air light is activated with on-air microphones, headphone output #1 is muted.

4.2.6 LAND PHONE LINE

6 The MX2200 has an RJ11 connector for connecting to the old terrestrial telephone line, also known as POTS (Plain Old Telephone Service). The NULL preset is used to adjust the rejection factor of the hybrid. A phone set can be connected using a parallel derivation-adapter on the telephone jack socket.

The phone line can be connected directly to the public telephone exchange or to a private exchange (PBX). However, private exchanges may deteriorate the rejection of the hybrid somewhat, so it is recommended to connect the console directly to the public exchange lines, if possible.

To protect against high voltage discharges, the telephone lines are internally equipped with metal oxide varistors. However, it is recommended to have additional protection based on gas discharge tubes and fuses, especially in areas prone to storms.

⚠️ Although the console has internal protection for distant lightning strikes, lightning damage is NOT covered by the warranty.

4.2.6.1 Reject setting (Null)

It adjust the rejection factor on the MX2200 hybrid. The user only needs to perform the setting when connecting the console to the old phone line. It is not necessary to repeat it if the console is disconnected and then reconnected to the same telephone line. However, it will need to be reset each time you connect to a different phone line.

The rejection factor determines the ability of the hybrid to prevent the transmitted signal from being mixed with the signal of the distant person. The higher the rejection factor, the better the quality of the local announcer's voice.

To adjust the rejection factor, follow these steps:

1. Send spoken content to the PGM from a computer. If no spoken content is available, you can speak live using the microphones. Music can also be used, but it may make the adjustment more difficult.

2. Establish a telephone call using the landline phone input.

3. The person who answers the call should remain silent, or activate the MUTE function if available.

4. Release the PGM button in the CUE MONI-TOR zone to remove the PGM signal from the headphones.

5. Listen to the telephone line in CUE and turn the NULL preset knob to one side and the other until the headphone level of the signal being sent to the PGM is minimum.

6. End the call.

4.2.7 STEREO AUXILIARY INPUT

7 Unbalanced stereo line level input. It uses a ¼" TRS Jack connector.

4.2.8 PROGRAM INSERTION

8 The MX-2200 has a stereo input called "Stream-In," which uses a switched 1/4" TRS jack plug. When this input is connected, it replaces the internal PGM mix on streaming and USB. This feature allows you to connect an external audio processor to the PGM output and re-enter the processed signal to the MX-2200 via the "Stream-In" jack, so that the processed audio can be streamed.

4.2.9 PROGRAM OUTPUT

9 Stereo PGM Out is a stereo output with a ¼" Jack connector. It sends the mix of the microphones, the USB, the aux line and the telephone lines.

4.2.10 ETHERNET

10 Port for wired connection to the local network. It is used for:

- access the configuration of the CODEC MX-2200 module via the web.

- Provide Internet connection for SIP transmissions and streaming.

- access the recordings made by the console via the web.

The Ethernet port on the MX-2200 console can be used to connect it to a LAN using a standard UTP cable. By default, the console has Dynamic Host Configuration Protocol (DHCP) enabled, which means that the LAN router/switch will automatically assign it an IP address. This allows the console to be remotely configured and monitored from a computer on the network using a web browser.

4.2.11 USB PORTS (Wi-Fi and MODEM)

11 The USB 2.0 type A ports allows to connect the following accessories:

- The Wi-Fi antenna for connection to wireless networks

- 4G modem dongles for connection to cell phone networks (extension cable required).

ATTENTION
Use an USB extension cable to connect the 4G modem dongle. Some devices can cause audio interference when are connected directly to the back panel.

This will enable you to keep the modem separated from the console and reorient it if you receive a weak signal indication.. |

4.3 CONFIGURATION

The options related to the Internet links can be accessed in the following ways:

1. From the onboard screen, using the rotary command or encoder.

2. Remotely, connecting MX2200 to a network (Wi-Fi or wired) and accessing with a computer with a web browser.

WEB ACCESS
1. Connect the console to the local network using the Ethernet port (recommended) or the Wi-Fi antenna.

2. Go to the SETUP screen to visualize the IP address that was assigned (DHCP is enabled by default). The first three items show the connection status and the assigned IP address for Ethernet, Wi-Fi and the Modem.

3. Using a computer connected to the same network, open a web browser and enter the IP address of the MX 2200 to access the web interface. |

4.3.1 SETUP SCREEN

The SETUP screen is one of the four main screens that can be accessed by turning the encoder. It provides access to all network and streaming options.

```
SETUP
WiFi 192.168.1.131@Solidyne
Ethernet 192.168.0.127
Modem 0.0.0.0
```

The top three options in the SETUP menu are Wi-Fi, Ethernet, and Modem. The current IP address is displayed on the right, or an indication is given if the connection did not receive an IP address. In

the case of a Wi-Fi connection, the name of the network to which the device is connected is also displayed.

Pressing the encoder the user can access to all the SETUP options, which are detailed below.

4.3.2 SETUP→ Wi-Fi

The Wi-Fi option in the SETUP screen allows the selection of a network from a list of available 2.4G and 5.8G networks. The signal level, measured in dBmV, is displayed next to each network name. Networks are listed in descending order of signal strength.

The MAC address of the Wi-Fi interface is displayed in the upper right corner. Some Wi-Fi providers require the MAC address of the device to enable connection.

4.3.3 SETUP→ ETHERNET

```
SETUP/Ethernet
Hostname mx2200min
IP 192.168.0.118
MAC 00:1E:06:48:21:9F

DHCP No
Netmask 255.255.255.0
Gateway 192.168.0.1

Netmask 255.255.255.0
Gateway 192.168.0.1
DNS1 auto
```

This screen allows you to configure options for the Ethernet network connection, such as the name of the equipment (Host name), DHCP mode, among others.

By default the DHCP option is enabled. MX2200 receives a dynamic IP address automatically.

To work with a static IP address, disable the DHCP option and manually assign the IP address, subnet mask, gateway and DNS servers. This and other advanced configurations must be done by specialized technical personnel.

The MAC address of the equipment is included for information only and it is not possible or necessary to change it.

4.3.4 SETUP→ MODEM

MX2200 supports the connection of USB 4G MODEM dongles (subject to compatibility).

Modern dongles act as an Ethernet adapter and its operation is plug & play: after a few seconds of plugging it is already available to navigate. It is not necessary any configuration.

The old 3G MODEMs need to be configured with the APN, username and password parameters. Please consult your Internet provider for these values.

When a connection is active will display the current IP address and it will be different from the null address (0.0.0.0).

```
SETUP/Modem
APN igprs.claro.com.ar
User clarogprs
Password
```

WARNING

Avoid directly plugging the MODEM dongle to the MX2200 as some devices may cause audio interference. It is recommended to use a short USB extension cable to keep the MODEM away from the console.

Additionally, this cable provides the flexibility to reposition the MODEM in case of weak signals.

4.3.5 SETUP→ RTP STREAM

```
SETUP/rtp stream
Enabled No
Protocol rtp
Dest 192.168.0.15

Port 3030
Codec MPEG Layer 3
Quality 320
```

Allows to the user to set up the RTP streaming options, such as the destination IP address, the port and the audio codec (MP3, Opus, PCM). This and other advanced configurations must be done by specialized technical personnel.

4.3.6 SETUP→ HTTP STREAM

```
SETUP/http stream
Enabled No
Protocol Shoutcast2
Server 62.210.129.76

Port 9500
Path /stream
Name MX2200

Codec MPEG Layer 3
Password s3b4s:)
Quality 160 Kbps
```

Configuration of the streaming settings for Icecast and Shoutcast servers. This and other advanced configurations must be done by specialized technical personnel.

4.3.7 SETUP→ SIP ACCOUNT

```
SETUP/SIP account
id sip:sebastian@mx2200.onsip.c
registrar sip:mx2200.onsip.com
username mx2200
```

This screen is used to enter the SIP account information for the console, which only needs to be configured once. The required information is typically provided by the SIP service used.

At a minimum, two SIP accounts are required:

- *SIP address for the console; for example mx2200@sip.linphone.org*

- A SIP address for the Studios; to be configured in the VoIP software used to receive or generate calls. For example estudioRadio@sip.linphone.org.

> **ABOUT THE SIP ACCOUNTS**
>
> The SIP account service is provided by third parties. There are providers that offers free services (linphone.org) and there are also paid services (for more complex VoIP telephony implementations). Please refer to section **5- Technical Notes** for details about the SIP accounts.
>
> Click on the following link to download a quick guide to create and to configure Linphone SIP with MX2200:
>
> www.solidynepro.com/DW/MX2200_Linphone.pdf

4.3.7.1 Audio quality of SIP communications

The SIP account QUALITY option enables the user to select the desired audio quality for SIP calls. The available options are:

Mono VoIP 16 kbps - Mono wideband 32 kbps - Mono FM 64 kbps - Mono HiStudio 128kbps - Stereo Mid 128 kbps - Stereo HiStudio 256 kbps -

By default, the MX2200 will use Opus CODEC with the user's default quality setting when initiating a call. However, if the receiving device/software is not compatible or prioritizes a different CODEC, the communication may use a different CODEC. In such cases, the user's selected transmission quality may not be applicable, and the device will use an alternative CODEC that is compatible with the current conditions.

4.3.8 SETUP→ SIP CONTACTS

```
SETUP/Contacts
FM Trujui sip:fmtrujui@fmtruju
Infotec Onsip sip:infotec@mx220
Pablo Linphone sip:pablospinell
```

Displays the SIP contact book stored in MX2200.

The user can navigate the contacts menu by pressing the encoder. Pressing it again allows editing the name of the selected contact.

Contacts can be entered and managed from the remote WEB interface.

4.3.8.1 Manage SIP contacts

To add a SIP contact:

1. To access the remote interface of the MX2200, simply open a web browser on any computer connected to the same network as the console and enter the IP address of the MX2200 in the address bar of the browser. The IP address can be found on the SETUP screen of the console or by checking the list of connected devices on the router or switch. Once the IP address is entered, press enter and the login page of the remote interface will be displayed. From here, the user can access the various settings and controls of the MX2200.
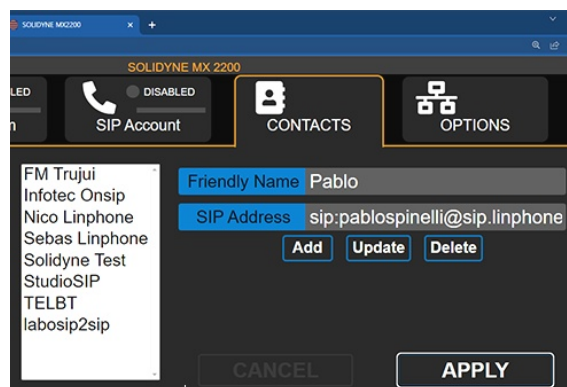
2. Click on the CONTACTS tab.

3. Enter the SIP address that you want to add in the SIP ADDRESS field.

4. Enter a short name for that address in the FRIENDLY NAME field.

   NOTE: If the fields contain text because a contact was previously selected from the list, simply overwrite the text with the new data.

5. Click the ADD button to add the contact to the address book.

6. Click the APPLY button to transfer the changes to the MX2200 hardware.

To edit a SIP contact:

1. Select the contact you want to modify from the list.

2. Edit the fields "Friendly Name" and/or "SIP Address"

3. Click UPDATE to update the contact in the address book.

4. Click the APPLY to transfer the changes to the MX2200 hardware.

To delete a SIP contact:

1. Select the contact you want to delete from the list.

2. Click DELETE. The contact will be removed from the address book.

3. Click APPLY to transfer the changes to the MX2200 hardware.

4.3.9 SETUP→ POWER INFO

The power display shows information about the power supply and battery charge voltage.

```
SETUP/Power
Power: 11.84 V External
Charging: dissabled
Bat level: ▮▮▮▮  100%
```

When the console is running on its internal battery, it will display a message along with the current battery voltage, such as "7.0V Battery". The "Bat level" indicator shows the remaining battery level, with a fully charged battery providing approximately 3 hours or more of streaming time. By default, the equipment will automatically turn off when the battery voltage drops to 6.2V.

An external 12V power bank can also be used to power the equipment.

4.3.10 SETUP→ TIME ZONE

```
TIME ZONE
GMT-02
GMT-03
GMT-04
```

The time zone setting is required for proper naming during recording. It is normally set only once. Time synchronization is done through public NTP servers.

4.3.11 SETUP→ RX VOLUME

```
SETUP/Rx Volume
0 ———————|——————— 200
          100
```

When the link is established via SIP, this option allows to set the level of the return signal.

The return signal is added to the CUE monitoring circuit (headphones) and to the POTS and Bluetooth sends.

4.4 Internal software update

Entering the IP address of the console with the subpath /update (example 192.168.0.121/update) accesses the web update page.

The software packages that comprise the system are displayed, along with any available updates. The most recent version is shown by default. Click 'Download' to download the update, and once the download is complete, click 'Install'.

This equipment supports updates via Wi-Fi. Software packages are downloaded to the device's internal memory, allowing the console to continue the update process autonomously in case of a dropped connection.

The final box displays hardware-related information, such as the number of battery charge cycles.
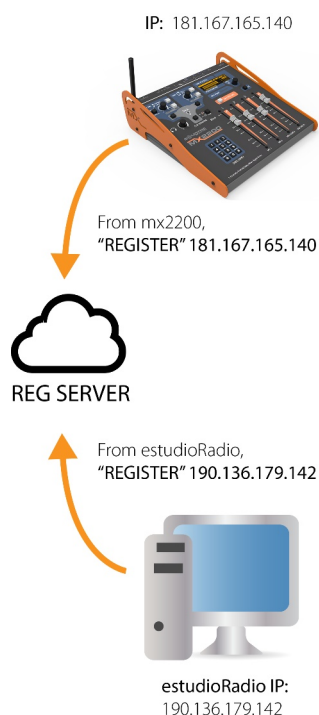
5 Technical notes

5.1 The problem of finding to the partner on the Internet

When two computers require communication with each other, it is imperative for both to be aware of the other's IP address. In certain cases, the user may opt for a static IP service to enable the remote team to easily locate the primary studio. Internet service providers often levy an additional charge for this service, and some providers may not even offer it, such as telephone cooperatives where the network is essentially a private subnet of the authentic internet (a private WAN).

Over time, several methods of establishing links have been developed, with the Session Initiation Protocol (SIP protocol) being a notable example.

5.2 What is SIP?

SIP (Session Initiation Protocol) is a communication protocol used by computers connected to the Internet to establish and manage multimedia sessions. It enables two or more endpoints, which may not know each other's IP addresses, to communicate by utilizing a cloud server.

IP: 181.167.165.140

From mx2200,
"REGISTER" 181.167.165.140

REG SERVER

From estudioRadio,
"REGISTER" 190.136.179.142

estudioRadio IP:
190.136.179.142

Each client on the SIP network is assigned a unique SIP address, which is similar in format to an email address, but is preceded by the SIP protocol indicator. For example, a SIP address may look like
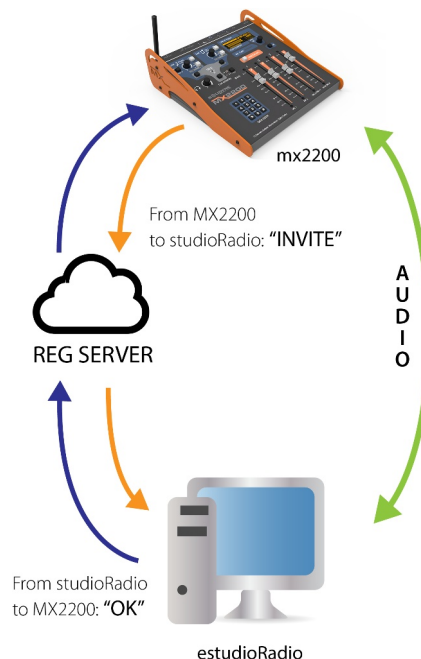
sip:studioRadio@sipserver.com.

To participate in the SIP network, each client must register with a SIP server, which maintains an updated list of IP addresses for all clients. The SIP server acts as an intermediary, allowing clients to discover and communicate with one another. However, once a connection is established, data is transmitted directly between client computers without going through the server.

Clients periodically update their information with the SIP server to indicate that they are still active and available at a certain IP address. Authentication is achieved through a combination of an username and password, which validates the client's identity to the server.
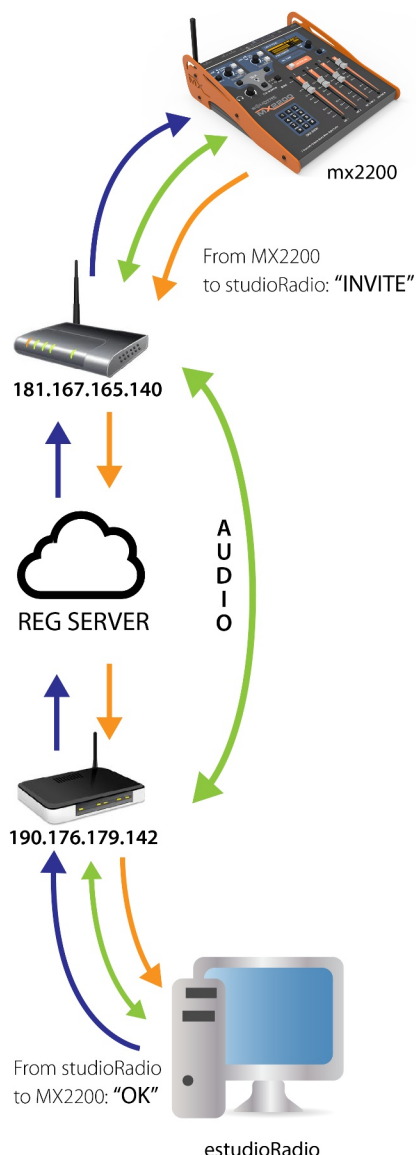
When one party, let's call it "A", needs to establish a link with the other, the following exchange takes place:

1. "A" sends an 'INVITE' message to SIP Server.

2. The server forwards the request from "A" to point "B".

3. "B" acknowledges ('RINGING') the server, which retransmits it to "A".

4. If "B" accepts the call, it sends an "OK" to the server, which retransmits to "A".

5. Both parties receive the IP address of each other to establish direct communication between peers. The session description protocol is included in the negotiation (SDP includes sample rate, channels, codec) to verify that both parties can work with the recognized audio format.

6. Audio (or text, images, video) is sent directly between the parties.

mx2200

From MX2200
to studioRadio: "INVITE"

A
U
D
I
O

REG SERVER

From studioRadio
to MX2200: "OK"

estudioRadio

When one of the parties decides to terminate the communication, a 'BYE' message is sent, and the other party responds with an 'OK' to confirm and end the communication."

In some cases, computers may be located behind a router or firewall, which requires the use of Network Address Translation (NAT) to establish communication between them.

mx2200

From MX2200 to studioRadio: "INVITE"

181.167.165.140

REG SERVER

AUDIO

190.176.179.142

From studioRadio to MX2200: "OK"

estudioRadio

If you want to know more about the SIP protocol, you can read the complete specification known as RFC 3261 'SIP: Session Initiation Protocol' published by the IETF (Internet Engineering Task Force).

https://datatracker.ietf.org/doc/html/rfc3261

https://es.wikipedia.org/wiki/Protocolo_de_iniciaci%C3%B3n_de_sesi%C3%B3n

5.2.1 How do I get a SIP account?

There are many SIP service providers on the Internet. Some are free and some are paid. Some have a mixed scheme, for example a number of free subscribers up to a certain number, others offer minutes of connection with traditional telephony for a monthly subscription.

When you get a SIP account the provider will give you the following connection details:

1) SIP address

In the form of sip:user@sipprovider.com

Some providers allow you to manage subdomains, for example if the station is called 'virtual radio', you can manage a subdomain and in this case your SIP address will be:

usuario@radiovirtual.proveedorSIP.com

Case differences in a SIP address are not considered.

By protocol, the prefix "sip: " must be specified so that it can be clearly differentiated from an e-mail address.

In some cases, the provider provides such disaggregated information, for example:

Username: luisgarcia

Server: proveedorSIP.com

And we must build the SIP address:

sip:luisgarcia@proveedorSIP.com

The SIP address (sip address) is also referred to by some providers as 'SIP identity', 'SIP URL' and also 'SIP URI'.

2) Domain

In the form of sip:xxx.supplier.com. For example:

sip:radiovirtual.linphone.org
sip:radiovirtual.onsip.com

Where Linphone and Onsip are SIP service providers. The domain (domain), is also known as 'URL of REGISTER' or also 'REG SERVER'.

3) Realm

Also known as 'Authentication realm'. It is a server that performs authentication. It can be equal to the domain.

Typically set to: "*"

What does the default connection server mean?

4) Username & Authentication username

In many systems it is allowed to differentiate the generic "username" from the "user used to authenticate". On other systems the username is simply ignored and only the authentication name is used (both are considered the same).

For example, the OnSip service (and others) allows to differentiate 'username' from 'Auth Username'. The Linphone service considers them synonymous.

In the exchange of messages between server and client, both data are sent at different stages of the process. The most important is obviously the 'Authentication username'; the simple username tends to have more descriptive effects.
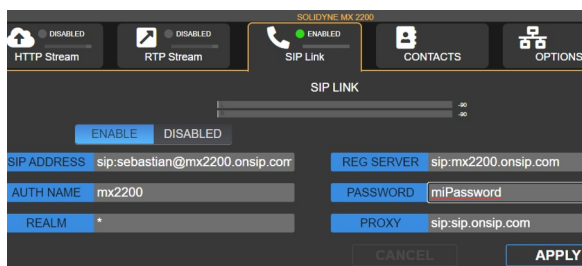
For example the 'Authentication username' can be 'lgarcia77' and the simple 'username' can be 'Luis_Garcia'.

If the provider tells you only one of the parameters then 'Authentication username' is equal to 'username'.

5) Password

Obviously, in the validation against the server, the password (password) assigned from the server will be sent to verify your identity.

SIP account configuration example:

To communicate with a contact, he only has to share his SIP address and you will add him to the contact book of the MX2200 (see Configuration section).

For example:

sip: estudioFM@radiovirtual.onsip.com
sip: studioproduction@linphone.org

Don't forget to include the sip: prefix to indicate the protocol. As with e-mail accounts, there is no restriction that a client registered with a SIP provider can send a call invitation to another registered with another provider.

5.3 Other definitions

IP adress

Every computer connected to the Internet has an IP address. It is basically a sequence of 4 numbers separated by dots and assigned/organized in a hierarchical manner.

Example: a (sub)net can be 192.168.0.x, all devices with such an IP address (example 192.168.0.100) can send messages to each other without using the external network.

The hierarchical organization allows the routing of messages (data packets) and thus discriminates when a message is destined for another node within the subnet or its destination is external to it.

Public IP address vs private IP address

The number of Internet addresses is finite, hence the need to organize internal sub-networks, also known as local area networks (LANs). Within the LAN, each device will have a local IP address, also called a private IP address, since it is not directly visible externally.

The gateway (GATEWAY) between the external Internet and the LAN is carried out by a device called a router, which is as-

signed a single external IP address or public IP and internally handles several internal or private IP addresses.

For security reasons, this equipment also acts as a "firewall" (FIREWALL) restricting external access to internal equipment.

What is NAT?

Typically, devices connected to a LAN receive an IP address that is internal to the LAN and belongs to a different family from the IP addresses of the real Internet. For this reason, when a computer wants to communicate with another device on the real Internet, it is necessary to use an automatic translation mechanism. This mechanism is known as a Network Address Translator, or simply NAT. It works by editing in real time the packets generated within the LAN and changing the internal source IP to the IP that the LAN owns externally. In this way the destination knows the (external) IP to which it will respond.

The response generated by the external server reaches the router, which modifies the source field before sending it to the internal network.

This allows multiple devices on the LAN to share the same external IP. The router is in charge of keeping a table with internal connections vs. external connections.

There are 3 basic classes of NAT and that can affect how SIP communication works.

RFC 2663: IP Network Address Translator (NAT) Terminology and Considerations

More info: https://datatracker.ietf.org/doc/html/rfc2663

What is STUN?

STUN (Session Traversal Utilities for NAT) is a client/server network protocol that allows NAT clients to find out its public IP address, the type of NAT it is in, and the Internet port associated with the local port through from NAT. This information is used to configure UDP communication between two hosts that are behind NAT routers. This protocol is defined in RFC 5389. The original specification, which became obsolete in October 2008, was defined in RFC 3489

What is TURN?

Traversal Using Relays over NAT (TURN) is a protocol that helps traverse network address translators (NATs) or firewalls for multimedia applications. It can be used with Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). It is most useful for clients on networks masked by symmetric NAT devices. TURN does not help running servers on well-known ports on the private network through a NAT; supports the connection of a user behind a NAT to a single peer, as in telephony, for example.

TURN is specified by RFC 8656. The TURN URI scheme is documented in RFC 7065.

What is ICE?

Interactive Connectivity Establishment is a protocol to allow two internal terminals on different LANs to communicate with each other as if they were on the same LAN.

RFC 8445 from the IETF:

Interactive Connectivity Establishment (ICE): A Protocol for Network Address Translator (NAT) Traversal

https://datatracker.ietf.org/doc/html/rfc8445

5.4 TECHNICAL SPECIFICATIONS

CODEC - LINK

SIP: Opus 48 kHz // Speex 32 K, 16 K, 8 K // G729 // G722

Streaming Icecast/Shoutcast: MP3 // Opus 64 to 320 Kbps

Streaming RTP: PCM 32 -44.1-48 kHz // MP3 64 to 320 Kbps // Opus 64 to 320 Kbps

CONNECTIVITY

Bluetooth 4.0

Gigabit Ethernet RJ45

WIFI 2.4G/5.8G USB (included)

4G Modem USB, external (not included)

AUDIO PERFORMANCE

Using RTP PCM 48 kHz = Frequency response 20 –18.000 Hz +/- 1 dB // Noise/Dynamic range > 85 dBA //

Distortion below 0,01 % THD

Analog MIC input using high performance EMI protected Opamps

Feedback controlled gain preamps: It never clips the audio signal at any fader position.

INPUTS

4 x MIC dynamic IN. Fader #3 has dual input preamp for MIC 3 + Mic 4 // Level = -70 to -30 dBu

1 x USB audio from PC at fader # 4 (when not using USB Fader#4 auto-switches to analog Stereo Line)

1 x Stereo Line, switched between Telephone/Line or Fader #4 // Level -10 / + 20 dBu

1 x Telephone Line to Hybrid – Standar POT line lines. VoIP Internet phone can be used with external FXS adapter, non-included

1 x Cellular Smartphone input using Bluetooth 4.0 (50 Hz - 8.000 Hz voice range quality)

PGM stereo insert for audio processor or external devices // Level + 4 dBu

OUTPUTS

Stereo Program Output: Analog; unbal; +4dBu
Max. level +14dBu; Z >5 Kohms

USB Stereo Program to PC

MIC mixer Out for Recording at PC// -10 dBu /10k

4 x Headphones Out (16-32-600 ohms)
Level + 5 dBu/50 ohms

Muted speakers out (replaces Headphone #1)

12 V / 120ma LED Tally Light

RECORDING

MP3 recording of stereo program.
More than 100 hours internal memory storage (@ 192 kbps).
16 bits / 48 KHz
Bit rate from 64 to 320 kbps
Nominal level -10 dBFS with 80 dB dynamic range.

INTERCOM

Internal talk-back MIC for private communication with Studio, Headphones, Telephone and Cellphone.

AUDIO PERFORMANCE

Frequency Response
MIC to PGM Out: 20 - 15.000 Hz +/- 1 dB Plus anti-pop -3 dB @ 30 Hz
                20 - 20.000 Hz +/- 2 dB
USB to PGM out: 20 - 20.000 Hz +/- 0,5 dB

NOISE:

MIC Equivalent input noise: -120 dBA // Line input S/N = 85 dBA

MIC COMPRESSOR:
Threshold 0 VU // Compression range: up to 20 dB over threshold
Attack time = 5 mS // Release = 200 mS

DISTORTION :
USB Line Input to USB PGM Out < 0,02 % THD at 1 Khz

Microphone at -50 dBu to analog PGM out < 0,1 % THD

PHONE HYBRID

Frequency response 300 – 3400 Hz +/- 1dB

Rejection: > 40 dB at 1 kHz with Null set at rear panel

Noise > 60 dBA S/N
On-board DTMF dialaer.

POWER SUPPLY

External Switching 90–240 V 50/60 HZ 12V / 1, 25A

Internal Litio – Ion battery 2 x 3,7 V / 2500 mA - internal charger

Battery operation time: about 4 hours (Wi-Fi on, Bluetooth off)

Optional external Power Bank: 12VDC /1000 mAh = 15 hr operation

INSERT EXTERNAL DEVICES

Stereo PGM insert for external processing of the streamed signal.
"STREAM IN" is a ¼" jack TRS with internal switch that disconnects (replaces) the internal PGM signal of the CODEC input (streaming) and the USB device. Input level +4 dBu.

DIMENSIONS AND WEIGHT

Wide= 260 mm // Deep= 284 mm

Front high= 30 mm // Rear High: = 95 mm

Weight = 2,4 Kg