



BLUETOOTH



# UNIDEX UX18

standalone & IP mixer console

OWNER'S MANUAL



# Summary

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# SECTION 1 OVERVIEW AND FEATURES

REVISION: March 2024

## 1.1 The concept behind "UNIDEX"

The Solidyne UNIDEX mixing consoles were designed to meet the needs of a wide range of radio stations.

The design premise was to create a digital console without the complexity of a '100% IP' console, which requires everything to be connected through a dedicated LAN, yet allows for great flexibility in connection and configuration, including networked audio.

### Simple adapters for simple installations

In addition to being the name of the line, UNIDEX is a special type of input that allows connecting external devices (UDX adapters) to handle mobile phones via Bluetooth, telecommunications, microphones, computers via USB, among other options. The UDX adapters connect directly to the console. Routers, switches, or network configuration are not required. The UNIDEX inputs manage a data channel that allows controlling the functions of each adapter from the console. These inputs can also be used to input conventional audio signals (balanced lines).

### Audio over IP for more advanced installations

The DANTE/AES67 connectivity simplifies the interconnection of devices and studios by sending audio over a data network, minimizing the number of cables and providing great flexibility in signal routing. For this, it is necessary to set up a dedicated network and have DANTE devices to input signals to the network (microphones, computers) and download signals from the network (monitoring).

UNIDEX is compatible with all DANTE devices on the market. The UX18-DANTE module is optional.

## 1.2 MAIN FEATURES

### 1.2.1 Overview

Solidyne UNIDEX UX18 is a standalone unit. It works at 24 bits @ 48 kHz with DSP processing.

Its compact profile presents all the inputs and outputs on the rear panel, using RJ45 (except the mic inputs which uses XLR). Solidyne can provide "RJ45 to audio-connector" patch cords (XLR, TRS, RCA) to resolve the installation without soldiers.

The startup is simple. UX18 connects and goes on-air with few previous settings. No advanced knowledge is required. Any operator with experience on broadcasting mixers can operate UX18 intuitively. The console does not differ from a conventional audio mixer for radio stations. Some features, such as monitoring assignments, sub-mix for groups and audio processing, are commanded from the touch screen.

There are **3 users profiles** that give access control. A basic operator can access only the minimum functions; but a senior operator can use advanced features and have access to all features and settings.

UX18 flexibility:

- Inputs can be assigned to any of the 6 physical faders, or the 12 "virtual" faders.
- The user can define sub-mixes, to group virtual channels and manage them from a physical fader.
- Until 12 scenes can be customized with different channel order or different processing settings (i.e. for the microphones).
- Remote access using a web interface.
- The computer connected by USB can manage communication software like Skype, Zoom, Google Meets, with mix-minus send and talk-back.
- From the computer connected by USB, real-time (low latency) VST effects can be used to obtain echo, reverberation, pitch and many other effects.

### 1.2.2 Channels

- UNIDEX UX18 mix engine manages 18 channels: 6 channels with 100 mm faders, 12 "virtual" channels commanded from the touch-screen (or remotely via web) and 4 Group Channels (that groups virtual channels).
- Each channel can receive audio from any input (inclusive from the *external tuner input*).
- Virtual Channels can be assigned to Groups (sub-mixes). This allows the user to mix virtual channels and to send a group on-air using fader/buttons.
- The channels can work in 3 modes: **MIC**; **LINE**; **TELCO**. These modes change functions and the behavior of the channel and sets according to the input.
- Channels can be assigned to 4 stereo mix buses: **PGM**, **REC**, **AUX**, **VBUS**.

### 1.2.3 Inputs

1. **5 balanced for microphones:** phantom powered (48V). 3 band parametric EQ, dynamic compression, noise gate and de-esser.
- **2 balanced lines** (stereo): with 3-band parametric EQ.
- **8 UNIDEX** (4 stereo): 8 balanced mono inputs. They can be used with the UNIDEX adapters to connect microphones, phone lines, cell phones (bluetooth) and other sources. UNIDEX inputs can be coupled in stereo pairs. They have 3-band parametric equalizer, dynamic compression and noise gate.

- **4 USB** (stereo): The computer connected by USB to the UX18 install four USB playback devices with supports to ASIO 3.0 These "USB inputs" (from the mixer point of view) have a 3-band parametric equalizer.
- **2 digitales AES-3** (estéreo): Supports S/PDIF signals. These inputs have a 3-band parametric equalizer.
- **1 incoming audio stream** (it is not an input itself because when activated it replaces the AES-3 #1 input).
- 1 input for an **external tuner** (unbalanced) Is available as a source in the MONITOR section, allowing you to listen to the on-air signal. This input can also be assigned as an audio source for the channels.

## 1.2.4 Outputs

There are 9 stereo outputs:

- **Double PROGRAM output** (fixed): Analog balanced and unbalanced.
- **1 salida REC** (fixed) balanced stereo.
- **4 unbalanced stereo** (assignable): Each stereo pair can be settled as independent-mono, to work with UDX external adapters. This outputs can be set to mix-minus to be used for communications.
- **4 USB stereo outputs** (assignable). At the computer side, are recognized by Windows as *Solidyne USB recording devices-1 to 4*.
- **2 digitales AES-3** (assignable).
- **Dual streaming** PCM/MP3/OPUS (RTP + Icecast/Shoucast *upstream*).

## 1.2.5 Monitoring and Intercom

- 2 outputs for studio monitors (loudspeakers + headphones).
- 2 outputs for control room monitors (loudspeakers + headphones)
- Talk-back to studio. Assignable between the built-in mic or an external mic (inputs MIC1 to MIC5).
- 2 Intercom lines assignable to outputs.
- Reverse-talk-back, to talk from the Studio to the Control Room (to the CUE circuit). Requires the optional module Studiobox HD3 or HD5.
- Bus CUE (PFL) (mono) assignable to outputs.

## 1.2.6 GPIO

UX18 has 2 GPI and 7 GPO, "Open collector". GPIO allows to command the Audicom software, digital audio processors or any other device equipped with remote control. Activation is programmable (see below).

## 1.2.7 StudioBox headphones splitter

The Solidyne StudioBox HD3 and HD5 accessories (optional) contains in a box all the monitoring connections needed into the Studio.



Image 1: Solidyne Studiobox HD3 y HD5

Features:

- Connection for 5 headphones in HD5 with independent controls of level (3 headphones in HD3 models).
- Double listening channel.
- Output for powered monitor speakers, with a volume knob.
- Tally light.
- Reverse talk-back, to speak from the Studio with the operator at CR (no matter the status of CUE in the microphone channels).
- Timer/Clock (only for HD5)

For details please refers to 3.6 - StudioBox HD3 & HD5

## 1.3 Technical support and upgrades

For specialized support and upgrades please contact to Solidyne SRL:

[soporte@solidyne.ar](mailto:soporte@solidyne.ar)

+54911 3119 3254 (Whatsapp only)

For videos tutorials about the use and configurations:

<https://www.youtube.com/SOLIDYNEbroadcast>

Diagrams, technical papers and application notes are available online in [www.solidyne.ar/unidex](http://www.solidyne.ar/unidex) (see details in Section 7 of this manual).

# SECTION 2 CONNECTIONS

## 2.1 Power on, off and resets

### 2.1.1 Power supply

The unit has an external switching power supply, 110/240V 50/60Hz.



Imagen 2 - Alimentación y llave de encendido

The power supply connector is located on the rear panel. First, connect the threaded connector to the rear panel, and then plug the power supply into the electrical outlet.

The unit does not have a power switch, so the UX18 powers on when the power supply is plugged into the electrical outlet. It is recommended that the electrical installation includes a dedicated cutoff switch."

### 2.1.2 Grounding

UX18 is grounded through the external power source. Do not use adapters or cables that suppress the grounding pin. It is also recommended to connect the chassis grounding terminal to the installation grounding, using cable with section 4 mm or higher.

The general electrical installation must be properly grounded. As reference, in Section 7 - DIAGRAMS AND APPENDICES you will find a grounding installation diagram for an FM station.

#### ATTENTION!

To avoid damage due to potential differences, make sure that the computer connected to UX18 via USB have a proper grounding.

### 2.1.3 Turning off the CPU

UX18 has an internal CPU that must be shutdown safely. The **POWER OFF** option is accessed from the main screen by tapping the '+' symbol in the STATUS section.

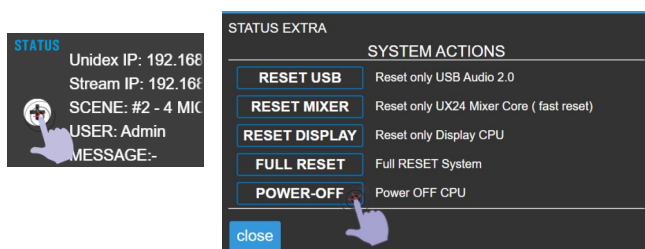


Image3: System actions

This action will not turn off the whole hardware. To turn off the surface disconnect the power source from AC outlet. Once the CPU turns off, wait a few seconds before turning off the power switch.

### 2.1.4 System actions

These functions appears into the option **STATUS "+"**. Allows to reset different stages of the mixer console. They are not used by the operator, but they can be needed in some cases for software/firmware updates.

**RESET USB:** Reset only the USB stage, forcing the operating system to recognize the device again.

**RESET MIXER:** Reset the mixing engine and the control surface.

**RESET DISPLAY:** Reset the computer, which manages the visual interface and the streaming.

**FULL RESET:** Full-reset of UX18 (is equivalent to turn off/on the unit).

## 2.2 About the RJ45 wiring

UNIDEX series come with inputs and outputs full connectorized with RJ45. And must be wired using CAT-5 shielded multipair cable (STP). With the advent of audio over IP (AoIP), several manufacturers began using RJ45 connectors and shielded multi-pair cable to replace the different audio connectors, standardizing the entire installation with a single type of connector and the same type of cable. The use of these connectors also allowed to considerably reduce the size of the consoles.

The use of structured cable facilitates installation anywhere in the world due to the availability of components and assembly tools used in data networks. At the end of the multipair cable the connection to an audio equipment (microphones, speakers, players, audio interfaces) will still require some type of conventional audio connector. For that, there are female RJ-45 patch cords to adapt to the audio connector needed.

### 2.2.1 Pre-assembled cables

Solidyne provides (optionally) all cables and adapters patch cords; intended to install the console without welding.

The following links details all available options:

<http://www.solidynepro.com/consola-de-audio-de-bc-acesorios-y-cables/>

[List of cables and adapters patch cords \(PDF\)](#)

## 2.3 Rear panel

The rear panel contains all the inputs and outputs. Most audio connectors use RJ45, with the exception of microphone inputs (XLR) and some others.

The inputs can be assigned to any of the 18 channels, physical or virtual.

The signal present on the outputs (with the exception of connectors PGM y REC) can be assigned from any of the

buses, or from some inputs in direct mode. The gain levels and other parameters of the I/O are configured from the touch screen (see SECTION 4 – Starting Up and routing).

For more details, please see the SECTION 4, Starting Up and Routing.

See the SECTION 7 on this manual for an infographic (online PDF) that summarizes the interconnection flexibility of the UNIDEX UX18.

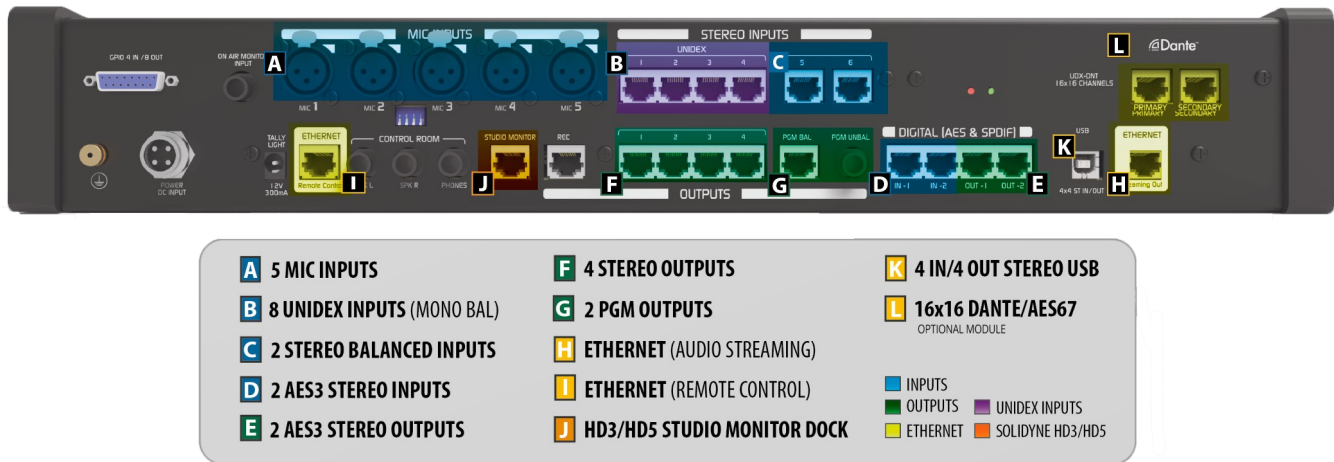


Image 4: Rear panel – inputs and outputs

### 2.3.1 Microphone inputs

**A** UX-18 has 5 microphone inputs, electronically balanced. All inputs gives switchable 48V phantom power. The connectors are standard XLR.

4 dip-switch allows the user to attenuate the preamplifiers gain in -12dB (moving the switch down reduces the gain). MIC-5 preamp gain is fixed.



Image 5: Gain attenuation

### 2.3.2 UNIDEX inputs

**B** UNIDEX inputs -a proprietary Solidyne development- are a type of **extended inputs that manages audio and other signals** (POE and communication) to control the Solidyne UDX external adapters.

There are 8 mono inputs, which can be used as stereo pairs (4 stereo). For the technical specifications UNIDEX inputs counts as mono inputs because they are used to connect mono sources as microphones, phone lines and mobile phones (Bluetooth).

The UDX adapters connects to the UNIDEX inputs via RJ45 to RJ45 patch cords (mod. SOL-P25 or SOL-P50). There are different UDX adapters:

- POTS [UDX 2TEL and UDX 2TEL-P adapters]
- Smartphones via Bluetooth [UDX TEL+BT]
- Microphones: Dynamic and phantom powered condenser [UDX 2MIC]
- Dynamic microphones [UDX DMIC]
- Computers via USB → AES3

UNIDEX inputs can also function as conventional line-level balanced inputs.

Each UNIDEX RJ45 contains two channels; left and right if stereo. In this case the connection is made using adapter cables by Solidyne (or others brands like Studio-Hub(c), by AngryAudio).

### 2.3.3 Balanced analog inputs

**C** 2 balanced stereo inputs. Each input uses one RJ45, since STP cable can carries both left and right balanced signals. At the end of the cable, the RJ45 is converted to an audio connector.

To connect an input in unbalanced mode, an adapter cable is used (SOL-M12/TRSF + adaptador TS a RCA; o SOL-M25/RCAM) The signal terminals (-) are connected to the cable shield, and the (+) terminal to the 'live'. See 2.5 - Connectors and wiring.

### 2.3.4 AES3 stereo digital inputs/outputs

**D** The UX18 has two AES3 digital stereo inputs. Each input uses one RJ45 connector.



AES3 inputs can be connected using cables Solidyne SOL-M12/1XLR or combining the patch SOL-P25 (RJ45-RJ45) with the adapter SOL-F3/XLRF.

AES3 use the same pin-out of the stereo balanced inputs, but only the left channel is connected. This inputs **support connecting S-PDIF signals**.

#### NOTE

When the **streaming download module is enabled**, the decoded audio **replaces AES3 #1** input and the rear panel connector is internally disconnected. The inputs AES1-L and AES1-R change its names to STREAM-L and STREAM-R.

**E** Two stereo AES3 outputs over two RJ45 connectors (balanced). AES3 connects using one RJ45-RJ45 CAT5 patch cord and the Solidyne adapter SOL-F3/XLRM; or SOL-M12/1XLRM. AES3 outputs implements the same pinout that the balanced stereo outputs, but only uses the left channel.

The signal present at the outputs AES3-1 and AES3-2 is assigned by the user from SETTINGS → OUTPUTS.

#### VERY IMPORTANT

**AES3 OUTPUTS ONLY WORKS AT 48 kHz**

### 2.3.5 Auxiliary input for external tuner

AIR MONITOR INPUT (TRS ¼" Jack) is a stereo line input available in the monitoring circuit. It can be assigned to the Studio and the Control Room.

In radio stations, it typically receives a signal from an external tuner to monitor the processed broadcast audio. However, it can also be assigned to any of the channels for use as an additional line input.

### 2.3.7 Assignable outputs

UX18 has 6 user-assignable stereo outputs:

- 4 analog (unbalanced).
- 2 digital AES3.

**F** The our analog outputs uses RJ45 connectors aligned with the four UNIDEX inputs, to simplify the connection of the UDX external adapters.

### 2.3.6 PROGRAM and REC outputs

**G** The outputs labeled as PGM always deliver the program bus mix. The signal is balanced on RJ45 and an unbalanced on TRS. The output labeled REC delivers the REC bus signal, balanced on RJ45."

### 2.3.9 USB audio "input and outputs"

**K** Built-in USB port provides 4 stereo "inputs" (*stereo playback devices* from the computer point of view); and 4 stereo "outputs" (*stereo recording devices* at from the computer point of view).

#### TAKE IN MIND

The USB port is a data transmission port, in this case used for transferring digital audio. In this manual, we refer to audio over USB as USB audio inputs and outputs to visualize it as traditional audio connections, where the console's 'USB outputs' are connected to 'USB inputs' on the computer, and vice versa. This is important for understanding the signal flow.

The signals send to the computer via USB are routing from SETTINGS → OUTPUTS.

#### NOTE

The following description refers to the built-in USB port. To install the external adapter **UDX USB** please refers to 2.11 – UNIDEX external adapters

The USB port requires a computer running Windows®10 or higher. Provides four stereo playback devices and four stereo recording devices. Requires the installation of a proprietary driver. To install follow the next steps:

1. Connect the USB port of UX18 to a USB port of the computer, using the USB A-B cable supplied with the console.

#### WARNING

The use of damaged or poor quality cables may cause the computer not to properly recognize the USB device.

2. Download the driver from the following link:  
[http://solidynePRO.com/DW/Solidyne\\_UsbAudio.zip](http://solidynePRO.com/DW/Solidyne_UsbAudio.zip)
3. Run the setup and follows the instructions.

Once installed, Windows® shows four Solidyne USB output devices, and four Solidyne USB input devices.

UX18 **supports ASIO 3.0**, allowing simultaneous shared playback between different software applications. For example, while an on-air play out software plays music over an USB device, other software takes an USB send from the console to add effects to the microphones in real time, using VST plug-ins.

The four **Solidyne USB playback devices** showed in the computer; corresponds to the inputs USB-1; USB-2; USB-3 and USB-4 in UX18.

The four **Solidyne USB recording devices** showed in the computer; correspond to the outputs USB-1, USB-2, USB-3 and USB-4 in UX18; which can be assigned to the buses (PGM, REC, AUX and VBUS) among other possible signals (see SETTINGS → OUTPUTS).

### 2.3.10 DANTE – AES67 (optional)

**L** DANTE module (cod. UX18/A67)

Gives 16 mono inputs (8 stereo) and 16 mono outputs (8 stereo). 4 stereo outputs are fixed to mix buses and 4 are assigned by the user.

UX18 STANDARD OUTPUTS	DANTE OUTPUTS
PGM-L	AoIP 01 (L)
PGM-R	AoIP 02 (R)
REC-L	AoIP 03 (L)
REC-R	AoIP 04 (R)
AUX-L	AoIP 05 (L)
AUX-R	AoIP 06 (R)
VBUS L	AoIP 07 (L)
VBUS R	AoIP 08 (R)
assignable	AoIP 09
assignable	AoIP 10
assignable	AoIP 11
assignable	AoIP 12
assignable	AoIP 13
assignable	AoIP 14
assignable	AoIP 15
assignable	AoIP 16

### 2.3.11 Audio streaming (upload/download)

**H** Ethernet port for streaming. UX18 can generate two independent audio streams, and can decode one incoming streaming.

The streaming with RTP protocol is commonly used to make a peer to peer link (for example to the transmitting plant -STL-). The Shoutcast/Icecast streaming is for upload to public servers.

Incoming streaming supports RTP and public http audio streams. When the incoming streaming option is enabled, the decoded audio is automatically switched to (replaces) the AES-1 input, that changes its name to STREAM-1

The audio streaming is done using the port ETHERNET Streaming Out.

**WARNING**  
ON THE REAR PANEL THERE ARE TWO ETHERNET PORTS: ONE FOR REMOTE CONTROL AND ANOTHER ONE FOR AUDIO STREAMING.

1. Connect "Streaming Out" to the LAN router.
2. The router will assign an IP address to the UX18 streaming module (DHCP). The IP address shows at the main screen as "Stream IP".

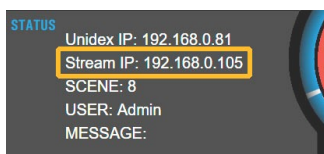


Image 6: Streaming services IP

3. The configuration can be done from the touch screen, or using a network terminal and a WEB browser and entering the Stream IP address.

### 2.3.12 WEB-control interface

**I** Ethernet connector for remote control. It connects UX18 to the LAN to enable the remote access via a web control interface, using a computer of the network. By default is in DHCP mode. The assigned IP is displayed on the main screen as UNIDEX IP.

### 2.3.13 Studio monitoring

**J** STUDIO MONITOR gives the signals for loudspeakers and headphones for the studio.

Solidyne StudioBox HD3 and HD5 monitoring boxes manages up to five headphones (HD5) and monitor speakers, among other features (for details see "3.5 - Studiobox HD3 and HD5"). It connects with a shielded multipair STP RJ45 to RJ45, flat pin-to-pin connection.

To use a standard headphones distributor/amplifier, an adapter patch cord is included with UX18.

#### CONNECTION WITHOUT SOLIDYNE STUDIOBOX

UX18 INCLUDES THE ADAPTER PATCH CORD "STUDIO MONITOR" (RJ45 to 2 JACK-TRS) TO CONNECT ACTIVE LOUDSPEAKERS AND HEADPHONES.  
**NEVER CONNECT THE "STUDIO MONITOR" OUTPUT USING OTHER RJ45-to-AUDIO ADAPTERS**

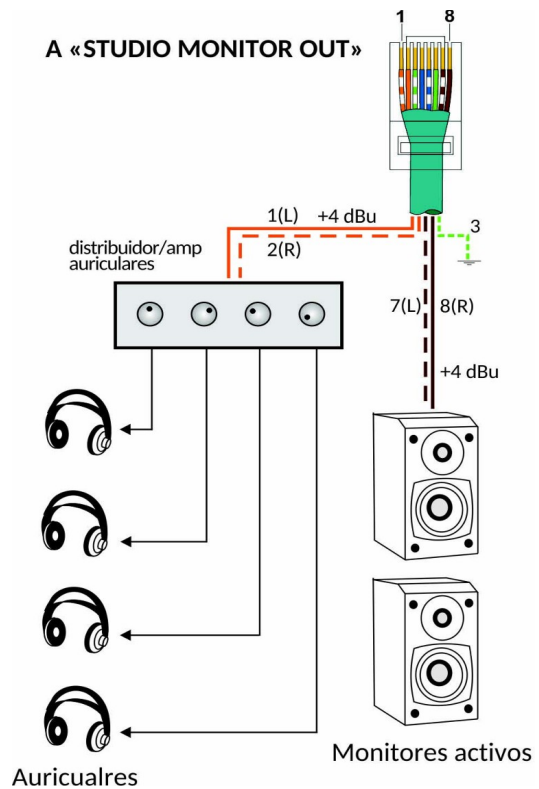


Image 7: Monitor speakers and headphones at the Studio, connected with a standard headphones amplifier.

#### STUDIO MONITOR pin-out:

RJ-45	STUDIO MONITOR OUT
1	Headphones, left (+4 dBu)
2	Headphones, right
3	GND
4	ON-AIR (+14 V)
5	Talkback to Control Room
6	VCC + 14 V
7	Speakers, left (+4 dBu)
8	Speakers, right

\*Chassis = GND

### 2.3.14 Control Room monitoring

In the radio stations that work with separate Control Room, the operator's headphones connects directly to the rear panel, to "CONTROL ROOM" (jack 1/4").

The monitor speakers of the Control Room are connected to the outputs "SPK L" and "SPK R" (jack 1/4").

On the mixer surface the operator have knobs to manage the volume of the headphones and the loudspeakers.

### 2.3.15 Tally light (ON-AIR signaling)

The output "on-air" gives 12 VDC / 250 mA when the mics turns on. This output supports direct connection of two Solidyne LA100 tally-lights. For other brands, make sure that the total consumption do not exceed 250 mA.

Optionally, GPO outputs 1 to 7 can be used to control individual on-air lights microphone arms. These GPO are of the "open dry" type (they switch between open circuit or ground) therefore they require an external power supply and, depending on consumption, a switching circuit.

**WARNING**

The outputs for tally lights are protected by a PTC or fusistor, which operates by over-heating when the current exceeds 250 mA; cutting the voltage at the output.

To restore the voltage, disconnect the load (tally light) and check the cause of over-consumption.

## 2.4 GPIO (connections)

The connector **GPIO** (General Purpose Input/Output) gives 2 inputs and 7 outputs. The GPI allows to command some UX18 features like mics activation. GPO can trigger external devices, such as "on-air" signaling lights; the Solidyne Audicom software or an audio processor.

- The GPI supports trigger signals from 5 to 15V / 150mA (active high).
- **GPO 1 to 7** are "open drain", supporting signals from 5 to 24 V / 250 mA.
- **Pin #8** gives fixed 15V/100 mA. It can be used to trigger a GPI (for example powering a relay). Or can be used to power a device switched by a GPO, but taking care that **never exceed the maximum supported current (100 mA)**.

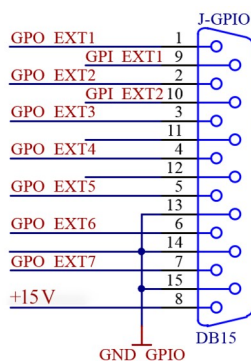


Image 8: GPIO pinout - female DB15

### 2.4.2 GPIO settings

The actions that controls the GPIO are set in the screen SETTINGS → GPIO. Please refers to 4.8 - GPIO settings.

### 2.4.3 Mic arms LEDs (GPO example)

GPOs 1 to 7 can work as electronic switches managing LEDs directly connected to an external power supply. The consumption must not exceed 250mA. To manage currents higher than 250 mA, use external electronic switches (relays or SSL circuits). Note that pin 8 can be used to feed a coil relay of 15V / 100 mA max. current.

In the example, the GPO are set to be active with the MIC channel on/off status (GPO1 → ON/OFF Channel MIC-1; GPO2 → ON/OFF Channel MIC-2; etc). The following figure shows an example of direct connection of a microphone arm with LED light, using GPO 1.

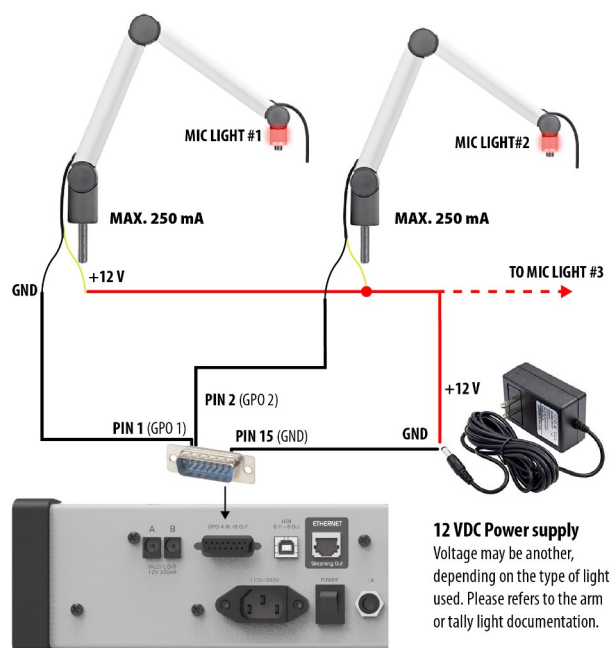
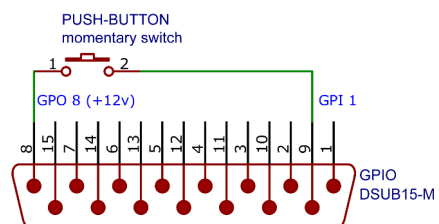


Image 9: Connection diagram of mic arms with tally light

### 2.4.4 Cough button (GPI example)

One action that GPI can manage is CHANNEL MUTING. This feature can be used to install a *cough button* into the Studio, in order to the talent/speaker can temporarily mute their microphone. The following example shows the connection of a push button to enable the GPI 1.



The pin 8 is fixed to 15V and GPI 1 is set to MUTE action, linked with the mic channel that corresponds (SETTINGS → GPIO). While the channel is muted, the operator can retake the control by moving the fader.

## 2.5 About the wiring

### 2.5.1 Set of cables and adapters

Solidyne offers a standard wiring kit (code MNG-UX18). Additionally, RJ45 to audio connector adapter cables are available for individual purchase. All available options are detailed in the following links:

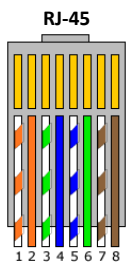
<http://www.solidynepro.com/consola-de-audio-de-bc-accesorios-y-cables/>

[Cables and audio adapters \(PDF\)](#)

Other brands cables such as the "StudioHub"(c) from *Angry Audio* can be used on all audio inputs/outputs; **with the exception of the output of STUDIO MONITOR** that uses a special cable supplied by Solidyne.

### 2.5.2 Making the cables

The following table shows the color nomenclature and standard pin numbering of an RJ45 connector.



RJ45 NOMENCLATURE	
PIN	WIRE COLOR
1	Orange/White
2	Orange
3	Green/White
4	Blue
5	Blue/White
6	Green
7	Brown/White
8	Brown

### 2.5.3 Audio over RJ45

#### BALANCED INPUTS / OUTPUTS

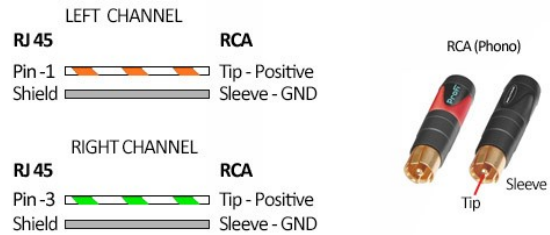
PIN	WIRE COLOR
1	Left channel (+)
2	Left channel (-)
3	Right channel (+)
4	Ground
5	Reserved
6	Right channel (-)
7	-15 (option)
8	+15 (option)

#### UNBALANCED INPUTS / OUTPUTS

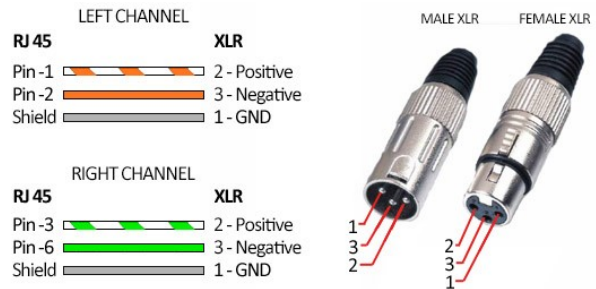
PIN	WIRE COLOR
1	Left channel (+)
2	NC
3	Right channel (+)
4	Ground
5	Reserved
6	NC
7	-15 (option)
8	+15 (option)

## 2.5.4 Standard audio connectors

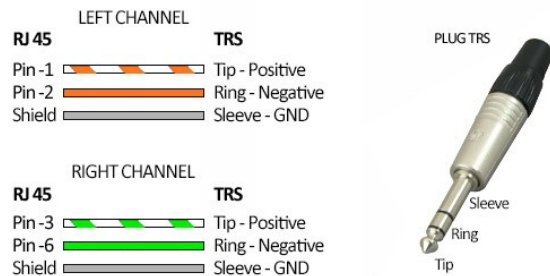
### 2.5.4.1 A RCA no balancedo



### 2.5.4.2 A XLR balanceado



### 2.5.4.3 A Plug TRS balanceado



### WARNING

The "SOL-M12/minijack" adapters (1/8" female TRS) can be connected to balanced or unbalanced outputs. It is not recommended to unbalance inputs, since leaves the negative (-) signal unconnected can pick up noise.

The "SOL-M12/RCAM" adapters (to RCA male) can be connected to balanced INPUTS to unbalance them, since they grounds the balanced negative (-) signal. **DO NOT USE ON BALANCED OUTPUTS.** To unbalance an output, leave the (-) unconnected.

# SECTION 3 SETTINGS

## 3.1 Starting sequence

When the console is turned on, the boot sequence loads the operative systems; the user interface and initializes the following settings:

1. The user profile that was defined for the start up.
2. The last time used scene.

### 3.1.1 User profiles and permissions

The user profile defines the access to certain features and settings screens. There are three users profiles, each one with different permissions:

#### BASIC - EXPERT - ADMIN

Default user is ADMIN, with full access. The profiles EXPERT and ADMIN are password protected, but the password never will be requested at startup.

In order to UX18 starts with restrict access, proceed:

1. Turn on UX18. The profile ADMIN will load (default).
2. Go to SETTINGS – USERS.
3. Change the factory DEFAULT USER “ADMIN” to BASIC or EXPERT.
4. Turn off the CPU and restart the mixer surface.
5. When the console restarts, the operator must know the password to change to EXPERT or to ADMIN profiles.

For more details please see 3.9 SETTINGS / USERS.

### 3.1.2 Inputs/outputs default routing

A scene saves the status of the inputs, outputs, bus routing, groups, audio processing and channel settings. There are 12 user scenes. Some scenes comes configured for different scenarios, to be used as example for other configurations. The user can edit the 12 scenes.

In order to facilitate the connection of the UDX external adapters, UX18 include some scenes configured to direct connection of (see the SCENE labels):

- One UDX-2TEL or UDX-TEL+BT
- One UDX-2MIC to add two microphones.

For more details please refers to 5.9 – Scenes and users profiles.

#### 3.1.2.1 Set to default (scene)

Is an internal read-only configuration. Touching the option SET TO DEFAULT, the current scene reverts to the default settings. This action requires to be logged as ADMIN. This action can not undo.

The following are the default settings for the scenes:

CH#	Group	Name	Mode	Input	M/ST
1	-	MIC-1	MIC	MIC 1	M
2	-	MIC-2	MIC	MIC 2	M
3	-	MIC-3	MIC	MIC 3	M
4	-	MIC-4	MIC	MIC 4	M
5	-	MIC-5	MIC	MIC 5	M
6	-	Bluetooth	TELCO	Internal Bluetooth	M
7	A	USB-01	LINE	USB1 LR	ST
8	B	USB-02	LINE	USB2 LR	ST
9	-	USB-03	LINE	USB3 LR	ST
10	-	USB-04	LINE	USB4 LR	ST
11	C	ST IN-5	LINE	STEREO IN 5	ST
12	-	ST IN-6	LINE	STEREO IN 6	ST
13	-	UNIDEX 1	LINE	UNIDEX 1 LR	ST
14	-	UNIDEX 2	LINE	UNIDEX 2 LR	ST
15	-	UNIDEX 3	LINE	UNIDEX 3 LR	ST
16	-	UNIDEX 4	LINE	UNIDEX 4 LR	ST
17	D	AES IN-1	LINE	AES1-L&R	ST
18		AES IN-2	LINE	AES2-L&R	ST

Output	Signal	Level at 0 VU**
OUT 1	REC	0 dBu
OUT 2	AUX	0 dBu
OUT 3	VBUS	0 dBu
OUT 4	CUE	0 dBu
USB1	PGM	-20 dBfs
USB2	REC	-20 dBfs
USB3	AUX	-20 dBfs
USB4	VBUS	-20 dBfs
AES1	PGM	-20 dBfs
AES2	PGM	-20 dBfs

\*\* Output level for 1 KHz at 0VU. With program signal, for a measurement of 0 VU, the peak level at the output will be higher, because the integration time of vu-meter.

## 3.2 THE SCREEN 'SETTINGS'

It access touching SETTINGS on the main screen.

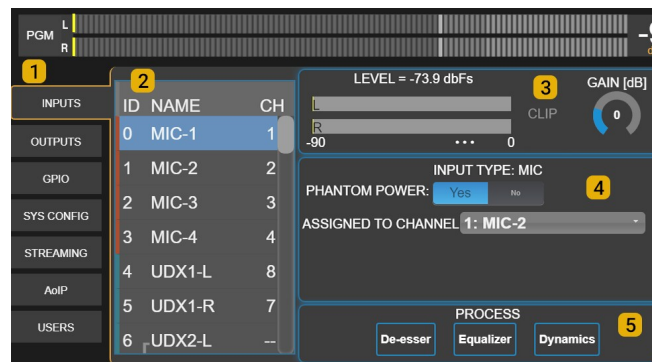


Image 10: SETTINGS INPUTS

**1 OPTIONS:** The settings options shows at the left menu. Touching an option, the right side will load the controls and values for each item.

### 3.3 SETTINGS: INPUTS

#### 2 UX18 INPUTS

List of all available audio inputs. To navigate the list, turn the encoder. Or touch an item to select it. At the right side, the options and settings of the selected input displays.

The ID showed at the right of each input is an internal fixed value.

The number showed at the right corresponds to the channel assigned to the input.

Stereo inputs shows as L/R pairs. For certain uses a stereo pair can be unlinked and be used as two independent MONO inputs.

#### 3 INPUT GAIN

Set the input gain. To edit the value, use the encoder. Push the encoder to switch between the list-navigation or gain value editing.

The level indicator shows the peak value in dBfs, post-gain (pre-processing and pre-fader).

The gain control is also accessible from CHANNEL screen. The gain adjustment can be made from any of these screens.

#### Quick guide to set the gain

1. On the channel that want to adjust, play high loudness music (i.e. mainstream pop); or speak to the mic with realistic "on-air" voice level.
2. Raise the fader to the nominal work zone (the zone marked between -10 y -20 dB).
3. Adjust the input GAIN until see that main VUmeter reaches 0 VU. Look at the slow indication, not the floating peak.

#### 4 INPUT FEATURES

The features that appears in this zone depends on the input that is currently selected.

**INPUT TYPE:** This information is present for all cases. Shows the input type: MIC, LINE, UDX, USB, AES.

**MODE:** Set the input as stereo or mono. When an input is used as mono, left and right channels works as independent channels.

**ASSIGNED TO CHANNEL:** Set the channel, from 1 to 18, to which the signal of that input is sent. The channel is chosen from the touch screen, using a drop-down list. This option is also available on the CHANNEL screen, from where it can also be edited.

**PHANTOM:** This option is shown only for MIC inputs. Turn on/off the 48V phantom power for that input.

**UDX TYPE:** This option is shown only for UNIDEX inputs. Set the type of external adapter connected to the input. The model is chosen using the touch screen from a drop-down list.

When a UNIDEX input is used as a balanced line input, UDX TYPE must be LINE (default).

#### 5 PROCESS

Shortcuts to the audio processes available in each input. In the usual operation, the processes are accessed directly from the options PROCESS or CHANNEL.

**6 SELECT:** In the screen INPUTS, the function SELECT toggle the buttons CUE/SELECT of hardware faders to the mode SELECT, to quick access to the input assigned to a fader (or channels assigned to a group).

### 3.4 SETTINGS: OUTPUTS

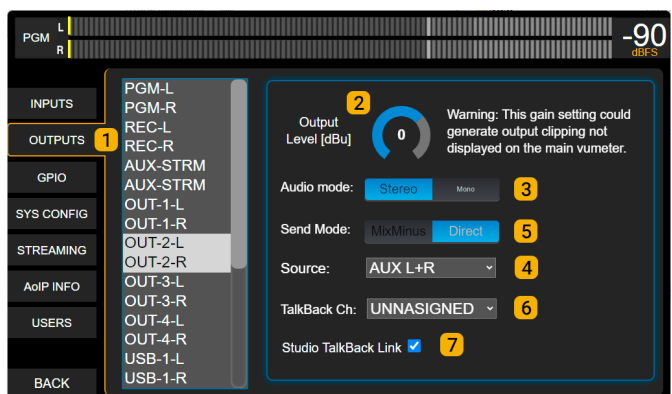


Image 11 - SETTINGS OUTPUTS

#### 1 UX18 OUTPUTS

List of all available audio outputs. To navigate the list, turn the encoder. Or touch an item to select it. At the right side, the options and settings of the selected output displays.

#### 2 OUTPUT LEVEL

Adjusts the output level. Pressing the encoder switches between the navigation of the list and edition of the gain value.

For **analog outputs**, the level is expressed in dBU referred to 0VU peak (default level + 4dBU).

For **digital outputs**, the gain is expressed in dB. The nominal output level (gain 0) is -20 dBfs referred to 0VU (1KHz @ 0VU = -20 dBfs).

#### About the level meters

The meters PGM, REC and AUX shows the signal level **previous to the output gain** (the BUS level). If a digital output gain is different to zero, the indication of the meters do not match with the real output level (dBFS).

**WARNING!** If the digital gain is positive, clipping can occurs at the output. The user will not be noticed by meters.

### 3 AUDIO MODE

Switch the output mode between mono or stereo. Each outputs can be used as a stereo pair, or as two independent mono outputs (for example, external adapters UDX 2TEL use mono outputs for sends).

### 4 SOURCE

Only available for assignable outputs. Sets the signal source for the output. Sources can be: PGM, REC, AUX, VBUS; CUE; STUDIO HEADPHONES; TALKBACK; MIC-1; MIC-2; MIC-3; MIC-4 and UDX IN 1 to 4.

### 5 SEND MODE

Set the output mode as DIRECT or MIX-MINUS.

**DIRECT** is normal mode for the outputs. The MIX-MINUS is disabled. The signal assigned at field SOURCE, sends directly to the output.

**MIX-MINUS:** When an output works in MIX-MINUS, certain input signal are not mixed to this output.

#### 3.4.1 MIX MINUS

A MIX-MINUS output is a special mix in which the signal of a given input is SUBJECTED from the mix. An application case is the audio send to a telephone line or communication software. If the signal sent (the signal that hears the caller) contains, for example, the PGM signal, this PGM mix should not contain the incoming audio from the communication, because it would generate a feedback loop (the communication incoming audio would be re-injected to the communication). Instead PGM, a special mix known as "MIX-MINUS" of PGM is sent; that is, a PGM mix that not include the incoming communication signal.

Outputs 1 to 4 and USB outputs can work as MIX-MINUS or normal (DIRECT) mode. This outputs shows additional settings options:

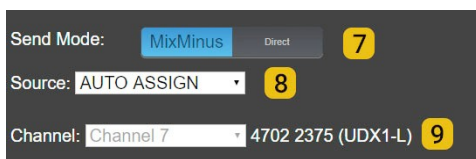


Image 12: MIX-MINUS settings

### 8 MIX MINUS SOURCE:

Define the BUS signal used to generate the MIX-MINUS. The options are: PGM, REC, AUX; VBUS or AUTO ASSIGN.

"AUTO ASSIGN" dynamically changes the BUS, according to the assignment made on the channel. If the channel is assigned to two or more buses, a fixed priority order define the BUS as follows:

- 1-PGM; 2-REC; 3-AUX; 4-VBUS.

For example, if the channel is assigned to all buses, the mix-minus is done against PGM signal, but if assigns to REC and AUX, the mix-minus is done against REC.

### 9 CHANNEL (mix minus mode):

Shows the input associated with the currently selected output, and the channel assigned to that input. This is a read-only field.

In order for an output can be used with MIX-MINUS mode, the associated input must be assigned to a channel. If the input is not routed, the MIX-MINUS option remains disabled.

The input associated to mix-minus is prefixed and cannot be changed. The following tables shows the default input assignment for the mix-minus:

OUTPUT	ASSOCIATED INPUT
OUT1-L	UDX1-L
OUT1-R	UDX1-R
OUT2-L	UDX2-L
OUT2-R	UDX2-R
OUT3-L	UDX3-L
OUT3-R	UDX3-R
OUT4-L	UDX4-L
OUT4-R	UDX4-R
USB OUT1-L	USB IN1-L
USB OUT1-R	USB IN1-R
USB OUT2-L	USB IN2-L
USB OUT2-R	USB IN2-R
USB OUT3-L	USB IN3-L
USB OUT3-R	USB IN3-R
USB OUT4-L	USB IN4-L
USB OUT4-R	USB IN4-R

## 3.5 GPIO settings

UX18 have 4 GPI (general purpose inputs) and 8 GPO (general purpose outputs). Connections and application cases are explained en 2.4 – GPIO.

The GPIO actions sets in the following screen:

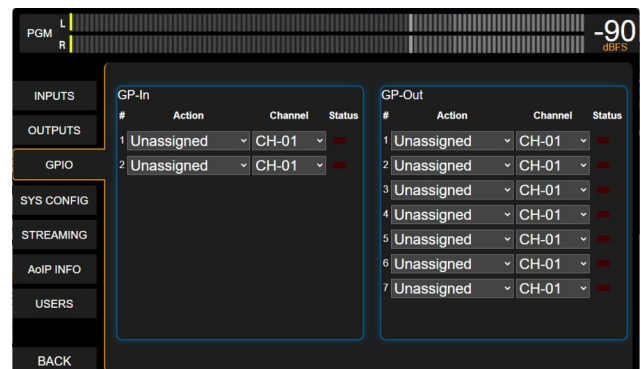


Image 13: GPIO settings

Each GP-In and GP-Out can be associated to a channel and to certain actions.

The actions available for the GPI are:

- UNASSIGNED
- CHANNEL ON/OFF
- CHANNEL MUTE
- MASTER MIC A
- MASTER MIC B

The actions available for the GPO are:

- UNASSIGNED
- CHANNEL ON/OFF
- TALLY A
- TALLY B
- TELCO RING
- OUTPUT +12V (solo en GPO8)

EXAMPLE: The following configuration:

GPI 1 | CHANNEL ON/OFF | CHANNEL 1

...will cause that Channel 1 turn on/off according to the signal present at the input GPI 1.

This configuration:

GPO 1 | CHANNEL ON/OFF | CHANNEL 6

...will cause that output GPO 1 changes its status when the Channel 6 is turned on or off.

## 3.6 SYS CONFIG

### 1 START ON-AIR INPUT

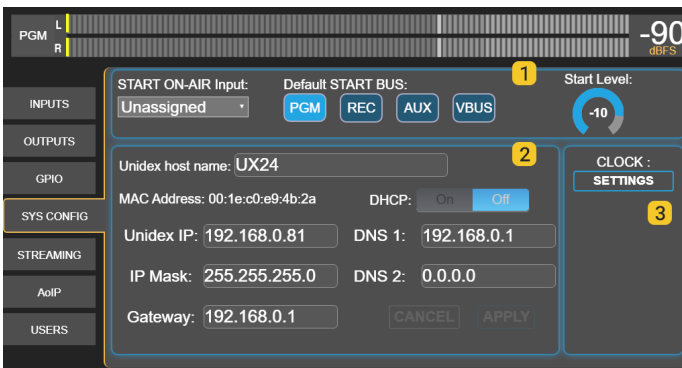


Image 14: System config

Set one input that will be aired in case the console restarts due to AC power failure. For stereo sources, select any input of the stereo pair.

The input must be assigned to a channel (hardware or virtual). The console always starts with the last time used scene. The audio source connected to that input must also be re-established by itself.

**Default BUS:** Buses that will be turned on for the channel associated with the “start on-air” input.

**Start Level:** Channel’s fader level. In physical channels, this level is independent of the position of the mechanical fader, but if the fader is moved, the level will change to the current position of the fader.

### 2 NETWORK SETTINGS (remote control)

LAN settings for remote control (Ethernet Control). By default the device works with DHCP enabled. The IP assigned by the router is shown on the main screen (UNIDEX IP at sector “STATUS”).

The IP address allows the user access to the console using a web browser. UX18 has an internal HTML5 web server, which allows full remote control to configuring and operating.

### 3 CLOCK SETTINGS

Shows a pop-up box to set the time. The clock can be settled in three ways:

1. Connecting UX18 to a LAN with Internet using Ethernet STREAMING port. In this case the sync is full automatic. No other action is required.
2. Connecting the UX18 to a LAN with Internet using Ethernet CONTROL port. In this case an internal clock UX-Core is synchronized. The user must sync manually the internal computer (wich controls the touch screen interface) by touching the option SYNC FROM UX-CORE showed on screen.
3. Manually from the screen. Used when the LAN do not have Internet access.

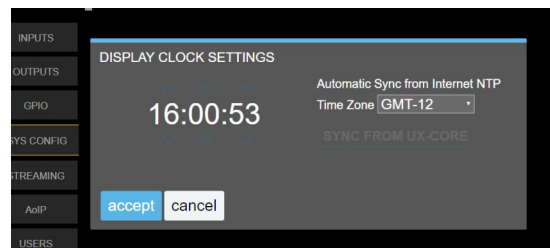


Image 15: Clock settings

#### NOTE

When UX24 access to Internet, it synchronizes with a World Clock (UTC). The user must define the Time Zone.

## 3.7 STREAMING

UX18 generates two independent audio streams and can decode an incoming stream.

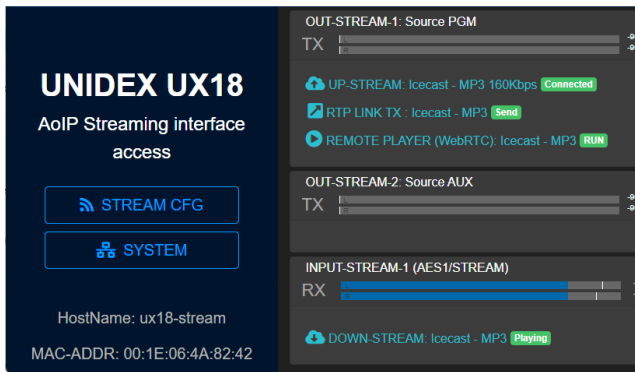
The options are:

- RTP Streaming for point-to-point linking.
- Streaming to Icecast/Shoutcast servers
- Incoming Streaming

Additionally, a WEBRTC service allows remote listening to the console’s audio from the WEB Control Interface, whether accessed locally from a LAN terminal or externally via the Internet.

Access to this section requires the ADMIN user. In the SETTINGS screen, the STREAMING option launches the Streaming Interface Access screen.





This screen provides an overview of the audio stream's transmission and reception status, and allows access to other configuration instances:

- **STREAM CFG** provides access to the streaming services configuration.
- **SYSTEM** provides access to the LAN configuration for the UX18 streaming services (not to be confused with the LAN configuration for WEB remote control).

### 3.7.1 UP-STREAM (TX)

The UP-STREAM tab presents the configuration options for uploading audio to Icecast and Showtcast media servers.

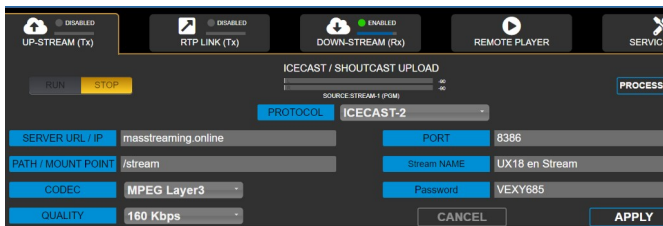


Image 16: Uploading audio stream for media servers

#### 3.7.1.1 Icecast/Shoutcast settings

- RUN/STOP:** Initiates or stops the upload to the server once changes are confirmed with APPLY. To optimize system resources, keep it stopped when the service is not in use.
- PROTOCOL:** Define if the server is Icecast or Shoutcast.
- SERVER IP:** IP address to which the streaming is sent. Is the address of Icecast/Shoutcast server.
- PORT:** TCP listening port for the server.
- MOUNTPPOINT:** A unique name that identify the stream at the Icecast/Shoutcast server.
- STREAM NAME:** Name that identify the stream.
- PASSWORD:** Password to access the server (if needed)
- CODEC:** Audio format for the transmission. Supports MP3 and OPUS coding.

- QUALITY:** Audio bitrate.
- APPLY:** Apply and save the settings and start/stop the service (according to RUN/STOP status)

#### NOTE

Settings for Icecast/shoutcast are given by the streaming provider.

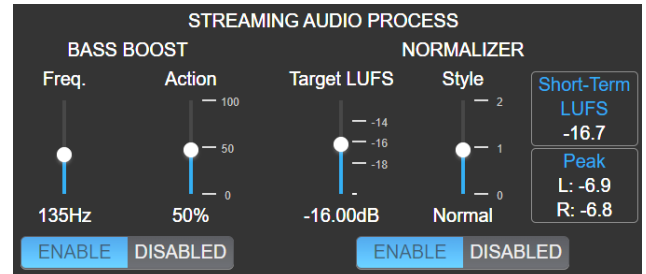


Image 17: Streaming audio Processor

#### 3.6.1.2 Audio processing for streaming

Touching the PROCESS icon provides access to an audio processor, which includes **level normalization** and **bass enhancement**. This processing only affects the streaming for Icecast/Shoutcast.

##### NORMALIZER

Reduces the signal dynamics and adjusts the level to a constant value, achieving a consistent streaming without volume jumps.

**TARGET LUFS:** Defines a target level for the streaming ( $\pm 1$  LU). It is adjusted based on the type of content predominant in the broadcast:

- From -21 to -19 LU for signals with a lot of level variation over which we have no control (for example, microphones without prior compression).
- From -18 to -15 LU for signals combining music and speech, always using microphone channel compressors to limit voice dynamics beforehand.
- From -14 to -12 LU for signals already heavily compressed (music).

**STYLE:** Defines the normalizer's action times among three possible values: slow; normal; fast.

As a general rule, it is recommended to use slow when the signal has large volume variations, while fast is suitable for normalizing already compressed signals.

##### BASS BOOST

Applies bass enhancement. The **FREQ.** control defines the cutoff frequency, below which the enhancement takes place. The **ACTION** control adjusts the degree of enhancement applied.

**BASS BOOST** is recommended for broadcasts focused on speech to achieve impactful bass voices.

### 3.7.2 RTP LINK

The RTP LINK tab presents configuration options to send streaming to a specific IP address. It is used to create peer-to-peer links over a LAN or via Internet. Some cases of use are as stereo link to a second Studio or as link to Transmitter Plant (STL).

**Implementing an RTP link requires advanced knowledge in network configuration.**

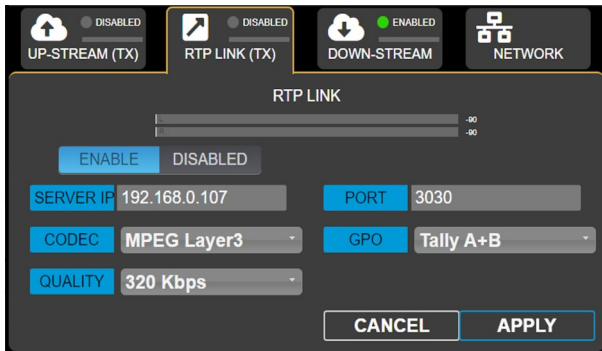


Image 18: Streaming RTP

- RUN/STOP:** Initiates or stops the RTP stream once changes are confirmed with APPLY. To optimize system resources, keep it stopped when the service is not in use.
- DEST IP:** Target IP address for the upstream.
- PORT:** Transmission TCP port.
- PLAYLOAD:** It's a numerical value related with the audio format. It is used by the decoder to recognize the audio stream.
- CODEC:** Audio format for RPT stream. Supports PCM, Opus and MP3.
- QUALITY:** Defines the bitrate for compressed formats (from 96 to 320 kbps) or the sample rate for PCM (32; 44.1; 48 kHz)
- GPO:** Sends the tally light signaling via TCP/IP to compatible Solidyne devices.

### 3.7.3 DOWN-STREAM (RX)

Allows to set up an incoming audio stream. When this service is activated, the decoded audio replaces the AES-1 input, which is renamed to STREAM-1 (the AES-1 connector on rear panel is disabled).

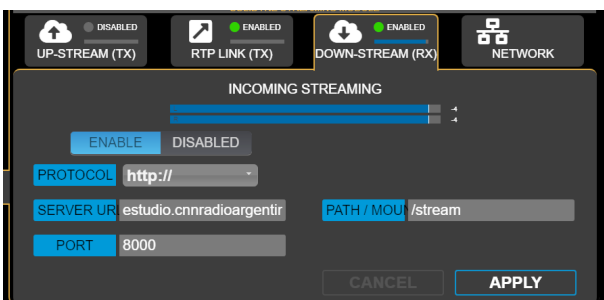


Image 19: Down-stream settings

**RUN/STOP:** Initiates or stops the down-stream service once changes are confirmed with APPLY. To optimize system resources, keep it stopped when the service is not in use.

**PROTOCOL:** Define the incoming stream format. The supported protocols are:

- Public streams HTTP and MMS encoded as PCM 16 44.1/48 kHz, MP3 or Opus.
- RTP streams PCM 16 44.1 kHz / MP3.

**SERVER URL/IP:** Source server name.

**PORT:** Incoming TCP port. By default is zero.

**MOUNTPPOINT:** A unique name that identify *the stream*.

**APPLY:** Apply and save the settings and start/stop the service (according to RUN/STOP status).

### 3.7.4 REMOTE PLAYER

The Web Remote Control interface (see SECTION 6 - WEB REMOTE CONTROL) allows listening to the console audio, which is sent through a WebRTC service.

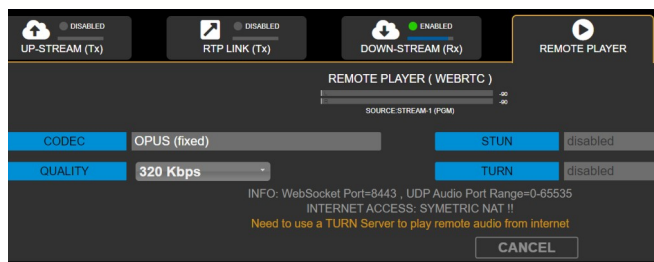


Image 20: WebRTC settings

The REMOTE PLAYER configuration screen displays information regarding the WebRTC service. The audio format is Opus. Users can adjust the bitrate within a range of 128 to 320 kbps.

When accessing the UX18's Web Control Interface from a local network terminal, no advanced configurations are required. The remote player will stream audio over the LAN.

However, when accessing the Web Control Interface via the internet, for the audio to be available on the remote interface, the network architecture must have symmetric NAT and a TURN server.

In a WebRTC communication implementation, symmetric NAT and a TURN server work together to enable real-time communication between devices, even when direct connection between them is not possible due to network restrictions.

### 3.7.5 SERVICES CFG

This screen allows assigning audio sources to streaming services. The options are:

- **STREAM-1:** Always delivers the Program signal present at the AES-1 PGM output.

- **STREAM-2:** Can be assigned to the PGM, REC, AUX, VBUS, STUDIO, CUE, or Talk-Back signals. It is configured in SETTINGS→OUTPUTS.

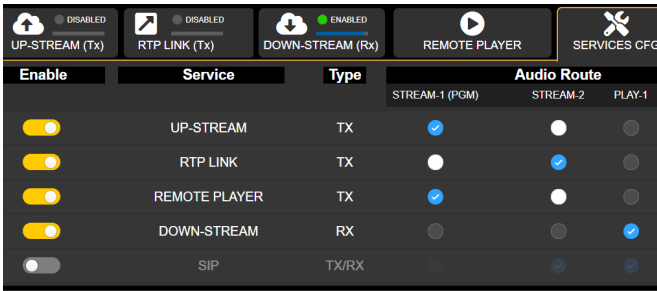


Image 21: Audio source for the streaming services

The audio gain for both cases is set from SETTINGS→OUTPUTS (AES1/STREAM1 and STREAM2).

The column **ENABLE** allows to start/stop each streaming service. For resource optimization, it is advisable to keep services that are not being used stopped.

### 3.7.6 STREAMING → SYSTEM

General network settings for the UX18 streamer module (do not confuse with the Ethernet access for remote control).

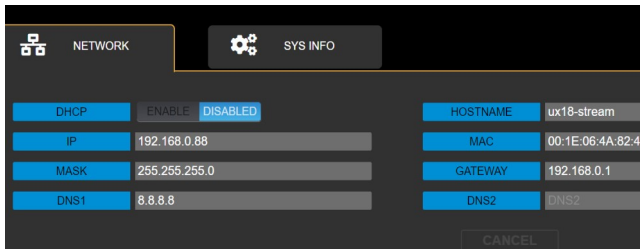


Image 22: LAN settings for the streaming module

By default, the IP address operates in dynamic mode (DHCP). The assigned address is displayed on the main screen as STREAM IP.

## 3.8 SETTINGS: USERS

UX18 has three user levels: ADMIN, EXPERT and BASIC. The users ADMIN and EXPERT are password protected. The passwords are set on this screen. Only the user ADMIN can set the passwords.

**Default passwords** are same user names, in lower case (*admin* and *expert*).

### DEFAULT USER

Defines the user loaded when the console starts. By default is ADMIN.

**When starting up, the system will never ask for a password, regardless of the default user set.**

To restrict access to advanced features, change the default boot user (ADMIN) to BASIC or EXPERT, depending on the level of restriction needed.

- If the unit starts up with the BASIC profile, the user will need to know the password to switch to EXPERT or ADMIN in order to access restricted features.
- If the unit starts up with the EXPERT profile, only features that require ADMIN credentials will be disabled.

## 3.9 AoIP

UNIDEX UX18 optionally can be equipped with the module DANTE (option UX18/A67) to obtain 16x16 networked audio channels.

AoIP INFO shows the settings and status of DANTE network.

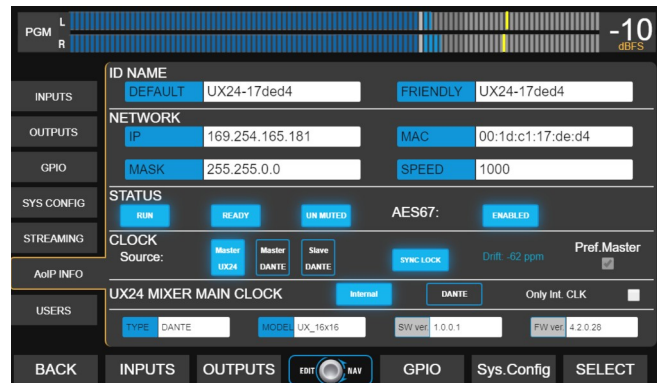


Image 23: DANTE-AES67

The audio flow in the network is configured using the DANTE CONTROLLER software developed by Audinet. Implementing this workflow requires knowledge of audio over networks.

Download from the web site an example diagram that shows a full implementation of UNIDEX with DANTE.

This document refers to UX24 but most considerations are valid for UX18. The PDF includes all Audinate's Dante links needed.

[http://www.solidynepro.com/descargas/productos/UNIDEX\\_UX24-A67\\_Dante\\_ESP.pdf](http://www.solidynepro.com/descargas/productos/UNIDEX_UX24-A67_Dante_ESP.pdf)

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# SECTION 4 UNIDEX EXTERNAL ADAPTERS

## 4.1 Overview

UNIDEX external adapters (UDX) allow you to expand the console's ability to manage many types of signals and devices. Each UDX adapter requires point-to-point connection to an UNIDEX input of UX-24, using RJ45 cables. Some UDX adapters also connects to UX24 outputs.

The operator controls the UNIDEX adapters functions from the console surface.

Available UDX adapters include:

- **UDX 2MIC:** Mic preamp of two inputs, 48V phantom powered.
- **UDX DMIC:** Mic preamp for dynamic microphone, built-in into a XLR connector.
- **UDX 2TEL y 2TEL-P:** 2-lines hybrid (POTS).
- **UDX TEL+BT:** Combine one POTS and one bluetooth link for smartphone.
- **UDX 2-BLT:** Double Bluetooth interfaces, to pair up to two cell phones.
- **UDX USB:** USB to AES-3. Stereo I/O to connect computers to the console.

## 4.2 UDX 2MIC

UDX 2MIC is a pre-amp for two microphones. Supports 48V "phantom" power, switchable from the mixer touch screen.



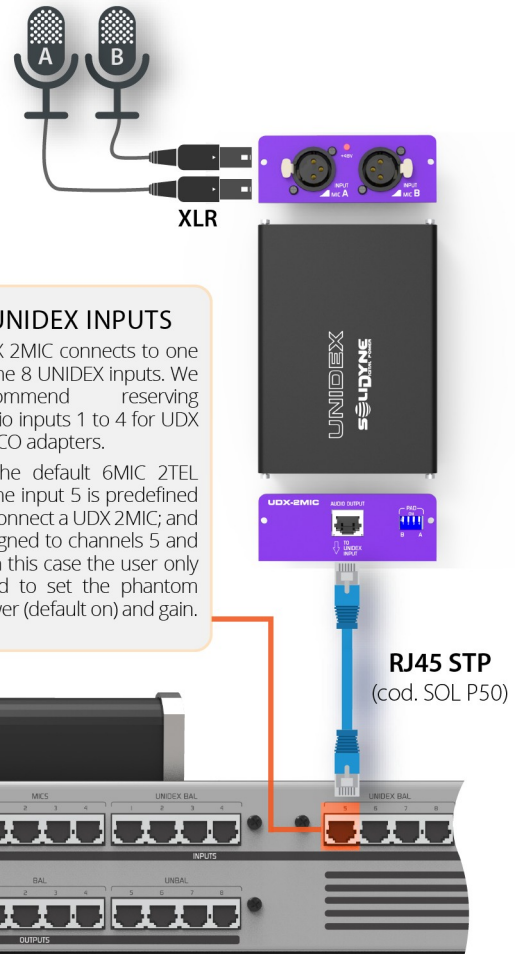
### 4.2.1 Setting up

UDX 2MIC is connected to a balanced UNIDEX input, using a single STP RJ45-RJ45 cable (Solidyne SOL-P50 or SOL-P25 cables).

Microphones connects to the UDX adapter using standard XLR audio cables.



UDX 2MIC has -12 dB pad dip-switches. The switches labeled A and B change the attenuation for inputs A and B respectively (the two central switches are unused).


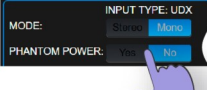
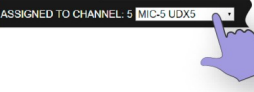



### 4.2.2 Set the input



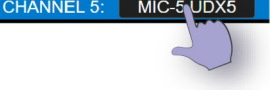
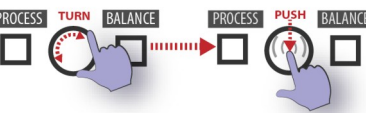

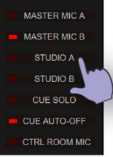
**a** SETTINGS SCENES PROCESS MONITOR TELCO CHANNEL  
(AT MAIN SCREEN)

**b** OUTPUTS TURN GPIO  
Turn the encoder until select the "L" input to which UDX 2MIC is connected. "L" corresponds to MIC A and "R" to MIC B.

**c** MODE: INPUT TYPE: UDX PHANTOM POWER: Yes No  
Set the input as MONO.

- d**  Set **UDX TYPE** as **2MIC**.
- e**  If needed, turn on the **48V phantom power**. It enables on both UDX2MIC inputs.
- f**  Assign the input to one of **CHANNELS**.
- g**  Press **BACK** to return to main screen.

### 4.2.3 Set the channel

- a**  Press **CHANNEL** on the main screen. All **CUE** buttons change its status to **SELECT**.
- b**  Press **SELECT** on the channel that wants to edit. **CUE** buttons returns to normal work and the main screen shows the **CHANNEL** options. To select a virtual channel, turn the encoder and push to confirm.
- c**  Set **MODE** as **MIC**.
- d**  To **edit the LABEL**, touch on text. A keyboard will appear.
- e**  Turn the encoder to choose **GAIN**. Push the encoder to edit the value.
- f**  Set the **GAIN** for peaks reaching -9dBfs at the level meter.
- g**  Configure the **ACTIONS** for the mic channel. The **MIC ACTIONS** allows: link the channel to the buttons **Master MIC A** or **B**; assign to **STUDIO A** or **B** (defines which "On Air" Tally Light will triggered); set the **CUE** mode; set as **MIC** at **Control Room** (defines which monitors will be muted).

To download a **Quick Installation Guide (pdf)** for this UDX adapter, go to **Section 7 – Diagrams and tech notes**.

## REMEMBER

**SAVE THE CURRENT SCENE TO STORE THE SETTINGS AND CONFIGURATIONS.**

## 4.3 UDX 2TEL and 2TEL-P

Modules UDX 2TEL and 2TEL-P allows to connect two POTS lines, with their respective phone sets.

All functions of **UDX-TEL** adapters are operated from the console surface (dial, answer a call, on-hold, dialogue privately). Each line is handled on individual channels.



Image 24: UDX 2TEL external adapter

The 2TEL-P model has on board controls that allow to answer calls with a phone set and transfer it to the console. In this way both the UDX and the associated telephones can be located away from the console, for example in a production office.

The 2TEL model has no controls on board. The calls are always answered by the operator from the console surface.

### 4.3.1 Connecting the UDX 2TEL/2TEL-P

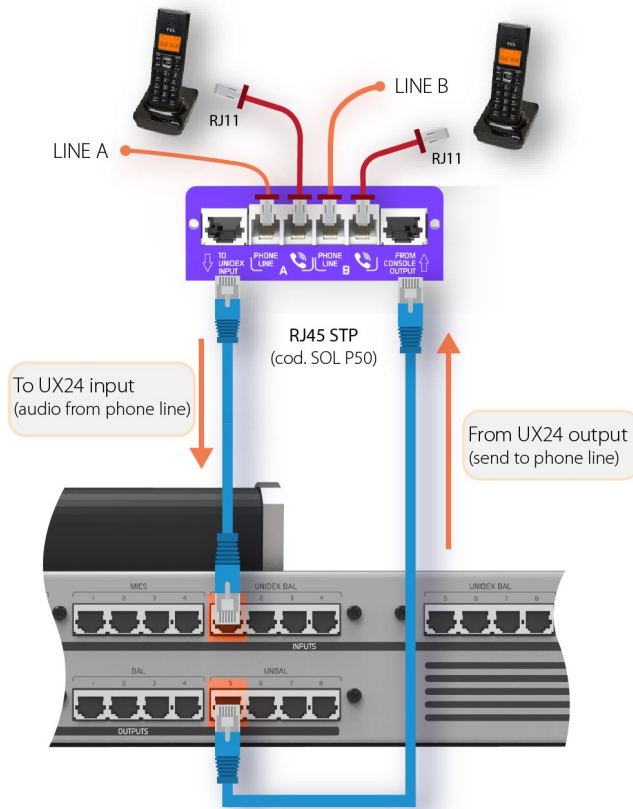


Image 25: UDX 2TEL – connectors

UDX 2TEL connects to the console using two standard STP cables with RJ45 at both ends (Solidyne SOL-P50 or SOL-P25)

- The output labeled "TO UNIDEX INPUT" connects to an UNIDEX input of the console.
- The input labeled "FROM UNIDEX OUTPUT" connects to an unbalanced analog output.

To simplify the connections for UDX-TEL, the UX24 rear panel aligns the UNIDEX inputs with the analog outputs.



### 4.3.2 Connection to phone lines

Telephone lines and associated phones are connected using standard RJ11 cables. The phones work normally while the lines are not taken by the console.

#### WARNING

If there is no dial tone when take a phone line, check if the LINE and PHONE wires are not crossed in the UDX TEL box

The lines can be connected directly to the public telephone exchange or to a private exchange (PBX). Private exchanges deteriorate the rejection of the hybrid, so we recommend connecting the adapter directly to the public exchange.

The UDX TEL adapters have internal RF rejection filters, both in the medium wave band (AM) and in the VHF and UHF bands. Telephone lines are internally protected against high voltage discharges (flashes or non-direct rays) by metal oxide varistors.

To prevent damages caused by voltage spikes, install additional external gaseous protectors over each line (see Section 7 – Grounding Diagram)

#### ¡ATENCIÓN!

Is strongly recommended to install additional Lightning/Surge Protector using Gas Discharge Tube (GDT) protection on each telephone line.

Once UDX-2TEL (or TEL+BT) is connected, proceed to set the inputs, outputs (sends) and the channels.

### 4.3.3 Set the inputs

- a SETTINGS SCENES PROCESS MONITOR TELCO CHANNEL
- b OUTPUTS TURN GPIO  
Turn the encoder to select the "L" channel at the input to which UDX 2TEL is connected.  
UDX-L = Line A  
UDX-R = Line B
- c INPUT TYPE: UDX  
MODE: Mono  
PHANTOM POWER: Yes No  
Set the INPUT MODE as MONO.
- d UDX TYPE: 2MIC  
Change UDX TYPE to 2TEL COMPACT.
- e ASSIGNED TO CHANNEL: MIC-5 UDX5  
Assigning a CHANNEL to the input.
- f PROCESS  
Equalizer Dynamics  
Equalization and noise gate can be pre-adjusted.
- g REPEAT THE PROCESS FROM ITEM (b) BUT CHOOSING THE CHANNEL RIGHT (UDX-R) TO SET THE LINE B

### 4.3.4 Set the outputs

- h BACK INPUTS OUTPUTS GPIO SYS SELECT
- i OUTPUTS TURN GPIO  
Turn the encoder to select the out "L" to which UDX 2TEL is connected.  
OUT-L = UDX 2TEL Line A  
OUT-R = UDX 2TEL Line B
- j Audio mode: Mono  
Set the output as MONO. To change touch the option on screen.
- k Send Mode: MixMinus  
Source: AUTO ASSIGN  
Channel: Channel 7 4702 2375 (UDX1-L)  
Touch to set Send Mode as MIX-MINUS.  
This way the output stays associated to the UNIDEX input that corresponds.

**l** REPEAT FROM ITEM (i) CHOOSING THE CHANNEL "RIGHT" (OUT-R) TO SET THE **LINE B**

**m**  Press **BACK** to return to the main screen.

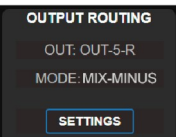
### 4.3.5 Set the channels

**a**  Press **CHANNEL** at the main screen. All the buttons CUE/SELECT change its status to **SELECT**.

**b**  Press the button **CUE/SELECT** on the channel that wants to edit. CUE buttons returns to normal work and the main screen shows the **CHANNEL** options. To select a virtual channel, turn the encoder and push to confirm.

**c**  Set **MODE** as **TELCO**.

**d**  To **edit the LABEL**, touch on text. A keyboard will appear.

**e**  Check if the output assignment indicates the corresponding output and if the mode is **MIX-MINUS**.

**f**  Press **BACK** to return to main screen.

### 4.3.6 ADJUSTING THE REJECT FACTOR

Only for POTS lines. The rejection factor or "null" factor must be adjusted for each line. Default factory is set to work with ideal line impedances; but in practice the telephone lines do not have the theoretical values, so a fine calibration is necessary.

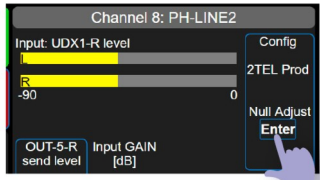
If the rejection value is not the optimal, when a call is on the air the voice of the Studio announcer can be affected (colored) by the audio that returns from of the phone line.


The rejection adjustment is needed for: UDX 2TEL; 2TEL-P and TEL+BT.

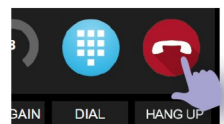
To make it proceed as follow:

**a**  Press **TELCO** to access to **TELCO** control panel.

**b**  **Make a call** (it can be an incoming calling or can be made from console). At the other end the caller must keep silence.

**c**  In the **TELCO CONFIG** section, press **Null Adjust ENTER**. The console sends a calibration tone to the phone line. The meter **INPUT LEVEL** shows the level at which the calibration signal returns.

**d**  The operator don't need to put the call on-air to make the calibration (main fader can be closed). The reject factor **adjusts from the UDX 2TEL box** using a knob on a preset, turning in both directions until finding the point where **the calibration signal that returns, shown on the meter, is the minimum**.

**e**  Once settled the reject factor, ends the calling. **The Null adjust must be set for each phone line.**

**WARNING**  
SAVE THE CURRENT SCENE TO STORE THE SETTINGS AND CONFIGURATIONS.

To download a [Quick Installation Guide \(pdf\)](#), go to **Section 7 – Diagrams and tech notes.**

### 4.4 UDX TEL+BT

The adapter UDX TEL+BT manages one standard phone line (POTS) and a **Bluetooth link** to pair a smartphone.



Image 26: UDX TEL+BT external adapter

The connection and settings for the POTS lines was detailed above. The procedure to pair a cell phone is explained in section 5.7 – **TELCO**.

To download a [Quick Installation Guide \(pdf\)](#), go to **Section 7 – Diagrams and tech notes**



## 4.5 UDX USB

This adapter allows to connect a computer, via USB, with one AES3 input and one AES3 output.



### 4.5.1 Set the AES input

- (IN THE MAIN SCREEN)
- Turn the encoder until select the **AES3 input** to which the UDX USB is connected.

INPUTS	ID	NAME	CH
	0	MIC-1	1
	1	MIC-2	2
	2	MIC-3	3
	3	MIC-4	4
	4	UDX1-L	8
	5	UDX1-R	7
	6	UDX2-L	
- Touch to set the **MODE** as **STEREO**.
- Touch the drop-down menu to **assign a channel**.

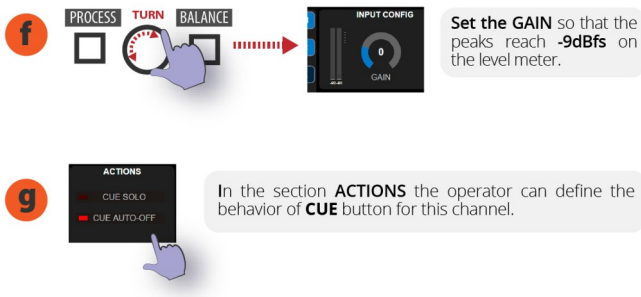
### 4.5.2 Set the AES output

- 
- Turn the encoder to select the **output channel** to which the UDX USB is connected.

OUTPUTS	PGM-L	PGM-R
	OUT-1-L	OUT-1-R
	OUT-2-L	OUT-2-R
	OUT-3-L	OUT-3-R
	OUT-4-L	OUT-4-R
	OUT-5-L	OUT-5-R
	OUT-6-L	OUT-6-R
- Set the output as **STEREO**
- SOURCE** sets the **signal source** for the output.
- Press **BACK** to return to the main screen.

### 4.5.3 Channel settings

- Press **CHANNEL** on the main screen. The **CUE** buttons change to **SELECT** mode.
- Press the button **SELECT (CUE)** on the channel that you want to edit. The console return to normal mode and shows the **CHANNEL** screen for the selected channel. To select **Virtual Channels**, turn the encoder and push it to confirm.
- Change the **CHANNEL MODE** to **LINE**
- To edit the **LABEL**, touch on the name. A touch keyboard will appear.
- Turn the encoder to focus the **GAIN** or touch the control. Push and turn to change the value.



## WARNING

THE AES3 OUTPUTS WORKS AT 48 KHZ.  
 SET THE WINDOWS RECORDING DEVICE TO 48 KHZ.

- In Windows10 go to System Settings and choose Sound.
- In Sound, go to Input > Device Properties.
- Open the option Additional device properties.
- Go to tab Advanced and set the sample rate to 48 kHz.

## 4.6 StudioBox HD3 and HD5: monitoring into the Studio

The modules for monitoring **Solidyne StudioBox HD5 and HD3** resolve the sending of all monitoring signals to the studio.



Image 27: Solidyne StudioBox HD5 and HD3

### 4.6.1 Connections

The HD3/HD5 accessories are connected to the **"Monitor Studio"** connector (UX24 rear panel), using a shielded multipair CAT-5 cable, with connection over RJ45 T568B (a 5 meter cable is included with the StudioBox).

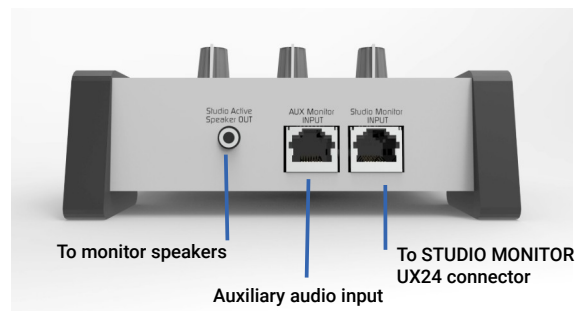
Through this cable the console sends the signals to the headphones and speakers; and receive the Talkback from Studio.

## WARNING

DO NOT CONNECT the OUTPUT Studiobox using RJ45 audio patch cords (it is not Studiohub compatible).



Image 28: Studiobox HD5



The **speaker output** uses stereo 1/8" TRS connector (mini-jack). This is a line level output, intended for use with **powered speakers**.

**Headphone outputs** use stereo 1/4" TRS connectors (jacks). Headphones with 32 or 16 Ohms of impedance can be connected (and combined).

## NOTE

For HD3, the headphones outputs 2 and 3 are twin: they shares the level knob and the source assignment.

The **aux input** is used to connect an alternative signal to the main assigned from the console (ON-AIR, PGM, AUD, SEND or CUE). The headphones can be switched between inputs A (console) and B (aux).

- On HD5, A/B switching is available independently for headphones 1, 2 and 3. Headphones 4 and 5 always receive the main "Studio Monitor" signal, assigned from the console.
- On HD3, the headphones 2 and 3 share the assignment switch.

The auxiliary input is unbalanced, with RJ45 connector. The connection to any RJ45 audio output of the console, balanced or unbalanced, is made using a standard STP RJ45 cable on both ends (SOL-P25 or SOL-P50 cables).

The UX18 console supports up to three HD3 or two HD5 units, which are connected on the "Studio Monitor" output in parallel. RJ45 type "Y" ("Y" splitter) junction boxes with pin-to-pin connection are used for the connection.



Image 29: RJ45 "splitter"

### Using headphones into the Studio

The Solidyne HD5 accessory manages up to five headphones with independent level knobs. HD3 allows to connect up to 3 headphones and has two level knobs. Headphones 2 and 3 are twins, sharing the level control. The maximum volume delivered is determined by the headphone level control in the MONITOR section of the UX24. What is heard depends on the signal selection made in the console.

### Use the loudspeakers into the Studio

The output for Studio speakers has a level control. StudioBox have dual A/B channel. Channel A is fed from the "Studio Monitor" connection, so the signal heard depends on the assignment on the console. Channel B receives the signal from the aux input (AUX Monitor Input).

**NOTE: SPEAKER MONITORS LEVEL**  
Set the volume using the speakers knobs for a comfortable listening, leaving the StudioBox LOUDSPEAKER LEVEL centered.

### About the Studio Intercom

The white buttons have two functions:

1. When the microphones are turned off, pressing the Studiobox Intercom the Studio can speak to the Control Room. Studiobox has a contact microphone for this purpose.  
At the Control Room, the operator will listen the signal by the console's CUE channel. The internal mic gain is adjusted with the preset "mic gain". In the console surface, the knob CUE level has no action on the HD3/HD5 mic signal, so that the Intercom will be heard even if the CUE volume is zero. The operator will use the talk-back as is usual to talk with the Studio.
2. When the mics are on the air, the Intercom button only works as a tally light.

**MIC GAIN**  
The preset located below the mic allows to set the mic gain and therefore, the listening level for the operator at the Control Room.

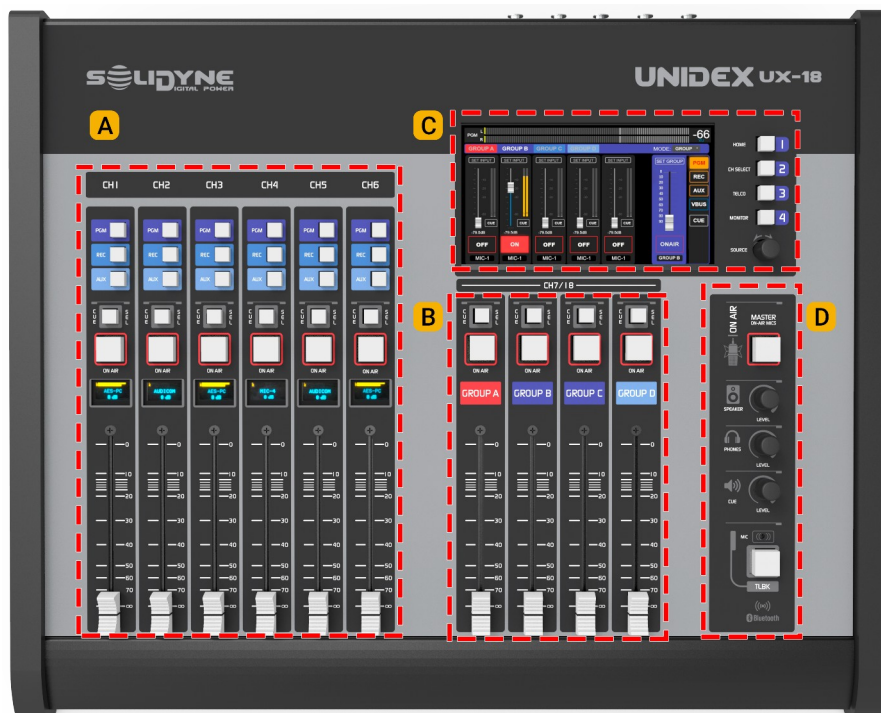
### About the Timer/Clock (only HD5)

Displays the current time (when the mics are off-air) or the elapsed time on-air (when the microphones are on).

The counter mode can be disabled. For this, remove the cover of the HD5 and remove an internal jumper located next to the clock. This will disable the counter feature, and the display will always show the current time.

## SECTION 5 OPERATION

### 5.1 Operating areas



**A 6 CHANNELS** presents traditional controls on the surface: main faders; out bus assign buttons, on/off and cue.

They operate in a traditional way and allow to manage POTS, VoIP and cell phones via Bluetooth (using external adapters UNIDEX TELCO).

**B 4 GROUPS** manages the channel's sub-mixes. Channels 7 to 18 can be assigned to the groups. Each sub-mix

are made using a touch-faders interface on the screen, that access pressing the button GROUPS.

**C 7" touch screen.** It has 4 direct access buttons at the right side and a push rotary encoder.

**D MONITORING** Monitor options for the Control Room and talk-back circuit. This area also contains the Master On-AIR MICS button.

## 5.2 MAIN SCREEN

### 1 LEVEL METERS

The level meters PROGRAM (PGM); REC; AUX and VBUS shows average level in VU (slow moving blue bar) and peak level (floating yellow segment) for the signals present in BUSES

The scale is in dBfs. **-20 dBfs corresponds to 0VU.** The level scale is valid for all meters. The numbers shown on the right indicates the current maximum level, which can be expressed in VU or dBfs. To switch it, tap the number on the screen.

The main meter remains visible over all screens and it can be switched to show other buses by tapping successively on the text "PGM".

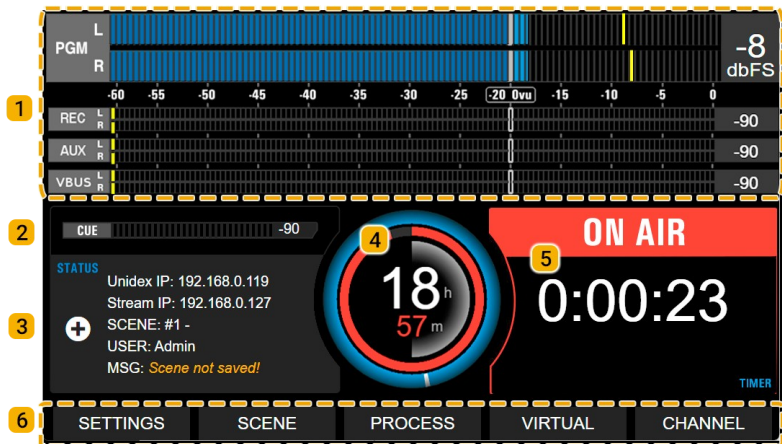


Image 30: Main screen

### LEVEL ON THE OUTPUTS AND METERS

The level meters show the signal level on the BUSES.

For digital outputs, the level indicator matches the level of the output if the output gain is zero. If the gain of an output is changed, the level at that output will not match that indicated on the meter.

Analog outputs are referred to 0VU = -20 dBFS with nominal level + 4dBu for balanced, and 0dBu for unbalanced. Setting an output level above the nominal level reduces headroom, and the output may hit clipping before the level meter shows it.

### 2 CUE LEVEL

Shows the level of the CUE signal. This level is posterior to the knob CUE Level.

### 3 STATUS

**Unidex IP:** IP address to access the UX24. Entering this IP into a web browser (Chrome is recommended) the users access to the remote control interface. The Ethernet port "Remote Control" must be connected to the LAN.

**Stream IP:** Destination IP address for streaming module. Entering this IP in a web browser, the user access to the configuration interface of the streaming services. The Ethernet port "Streaming" must be connected to the LAN.

**SCENE:** Shows the current scene.

**USER:** Shows the current user profile.

**MSG:** Shows system messages (for example when the scene changes are not saved).

**“+” ADVANCED:** It presents advanced options for system maintenance (see 2.1 – Power on/off and resets). The most important item is the **safety shutdown** of the internal CPU, which **must be done before a hardware power down.**

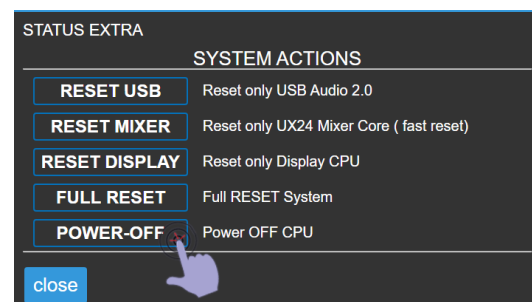


Image 31: System actions

### 4 CLOCK

Shows the current time. The time synchronizes via Internet, or can be settled manually. Please see 4.6 - SYS CONFIG.

## 5 ON-THE-AIR COUNTER

Indicates the time that the mics are on the air, showing an ascendant count in hours:minutes:seconds.

When the mics turns off, the elapsed time stays on screen.

## 6 MENÚ Y ACCESOS RÁPIDOS

Four operative screens are launched from hardware buttons, located at the right of the display. Other screens are opened from the main screen.

### DIRECT ACCESS BUTTONS

- HOME** Calls to the main screen.
- GROUPS** This screen shows the routing options and sub-mixes for the groups. Also include a BUS meter and the clock/counter.
- TELCO** Is the interface for manage telephone lines and Bluetooth linked cell phones. It presents all the necessary functions, including a keyboard to POTS dialing.
- MONITOR** Manages the monitoring signals for the Studio and the Control Room. Also contains advanced options related to headphones and monitor speakers.

### MENU ON THE MAIN SCREEN

- SETTINGS** Only the profile ADMIN has access. It contains all the advanced and system configuration options, including the assignment of inputs and outputs.
- SCENE** Allows to edit scenes and switch the user profile. Also contains the **CHANNEL MAP** that makes easier to re-assign channels and groups.
- PROCESS** Access the audio processors: parametric EQ, dynamic compression, de-esser and noise gate. The processes that are available depends of the type of input selected.
- VIRTUAL** Launch the on-screen mixer with the virtual channels. When a virtual channel is ON-AIR the icon VIRTUAL blinks in red.
- CHANNEL** This option shows the screen that summarizes the status of the selected channel. Gives access to essential controls such as the gain control and pan-pot; and to some advanced options such as audio processing and input routing.

### ABOUT SCREEN SAVER

All displays automatically turn off after 15 minutes of inactivity. Slightly moving a fader or pressing a button activates the displays.

## 5.3 Channels

UNIDEX UX18 manages 18 channels which can be assigned to any input (see 4.4 SETTINGS→INPUTS).

- 6 “hardware” channels (1 to 6) operates from the mixer surface, using traditional faders and mechanical buttons.
- 12 “virtual” channels (7 to 18) operates from a touch-screen mixer interface.
- 4 groups faders can manage each one a single virtual channel, or sub-mixes of two or more virtual channels.

All channels (hardware and virtualized) presents the following areas:

**1 BUSES:** Each channel can be assigned to 4 mixing buses: PGM; REC; AUX; VBUS

Channels 1 to 6 are assigned to PGM, REC, and AUX directly from the surface buttons. When the channel is assigned, the button lights up orange.

Channels 7 to 18 are assigned to output buses from the VIRTUAL CHANNELS screen (or from CHANNEL screen). When a channel is assigned to a group, it takes on the group's bus assignment.

For all channels, the assignment to VBUS is done from the CHANNEL screen in the main menu (also available in the virtual channels mixer).

**2 CUE/SELECT BUTTON:** It has three functions.

- CUE, pre-fader listening.
- SELECT, select the channel.
- TALKBACK to TELCO channels and private talkback to headphones.

In normal use works as CUE, to listen to the signal present on the channel, previous to the fader and the ON-AIR button.

The **SELECT** function is used to select a channel and perform some action, for example, set the gain, edit the label, etc. When this mode is activated the CUE buttons change to blue and the OLED indicates “SELECTED”. Details about this mode are explained later.

**TALK-BACK mode** is exclusive to TELCO channels, which manages telephone lines. By holding down CUE, the talkback microphone is enabled to talk to a phone line. Another use of this feature is to set a secondary private talkback send to one or more headphones into the Studio.



Imagen 32: Canal

**3 ON-AIR button: Turn on the channel.** When is ON, the signal sends on-air (to the mix buses). The level is set with the main fader.

**4 DISPLAY:** The channel OLED shows:

- The input level, previous to fader and processing. If the input signal is reaching 0 dBfs at the A/D converter, a **CLIP indication** appears. To solve this condition, turn down the source level or reduce the input gain.
- Name or label that identify the channel.
- Status or special modes like SELECT, TLKB or channel profiles like TELCO.

### OLED ENERGY SAVER

The brightness of OLED displays is **reduced after 5 minutes** of inactivity. **After 15 minutes, OLEDs are turned off.** Any action on the channel controls or Intercom buttons restores the displays.

**5 MAIN FADER:** Sets the level of the signal for the mixing buses.

The level adjusted by the fader on the surface can be changed from the touch screen (CHANNEL) or externally from the web interface. When this happens, the current attenuation value will not match the fader position, because the **faders are not motorized**. This condition will be indicated on the OLED screen, which will flash showing the current level value in dB.

When the fader on surface moves, the value is progressively updated to the current fader position.

## 5.4 Groups

UX18 present four group faders. Each group can have assigned one or more virtual channels (ch 7 to 18).

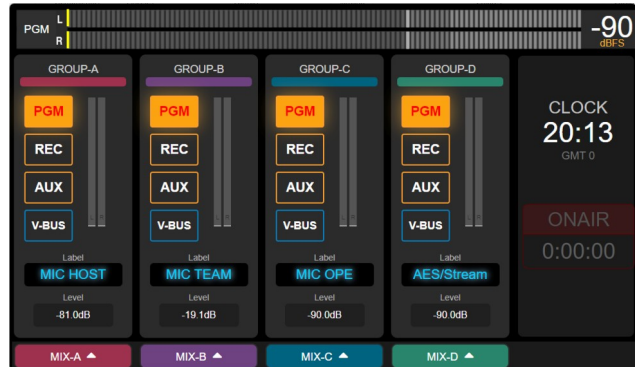


- When a group is assigned only one channel, it behaves like an individual channel, providing physical controls (fader, on/off button, and CUE) to the virtual channel.
- Groups works as sub-masters, handling sub-mixes of virtual channels.
- Group channels can be linked to the MASTER ON-AIR MIC button (from the CHANNEL screen).

Groups feature mechanical controls for level adjustment, on-off and CUE, while assignments to buses and sub-mixes are made from the touchscreen by pressing the GROUPS quick access button.

The GROUPS screen initially displays:

- **Bus assignment buttons:** defining the output assignment for each group. All channels linked to the group inherit this assignment.
- **Essential information:** allowing the operator to work with the GROUPS screen while keeping track of program level, current time, time counter, and indication of "on-air" microphones.



By touching the MIX option at the bottom of the group channel, a **mixer interface** is displayed showing the channels assigned to that group. These controls allow for configuring the subgroup mix. Group mixes are saved within the scene.

### 5.4.1 Assign channels to groups

Virtual channels can be assigned to a group in two ways:

- Through the **CHANNEL MAP** wizard, accessed from the SCENES screen or by pressing **ADD** in the group mixer view.
- From the CHANNEL screen, accessed from the main screen.

To assign or remove channels from a group, accessing with an ADMIN profile is required. Group settings are saved within the scene.

## 5.4 Monitoring section

The monitoring section consists of the following areas:

- Level knobs for loudspeakers, headphones and CUE located on the mixer surface.
- The screen MONITOR to assign the audio sources.

### 5.4.1 The screen "MONITOR"

The MONITOR function is accessed by pressing the quick access button located next to the screen. On this screen, the operator assigns the audio sources that will be heard through the studio's monitor speakers and headphones, as well as in the control room.



Audio assignment takes place immediately upon tapping an option. The selected option is highlighted as active in light blue color.

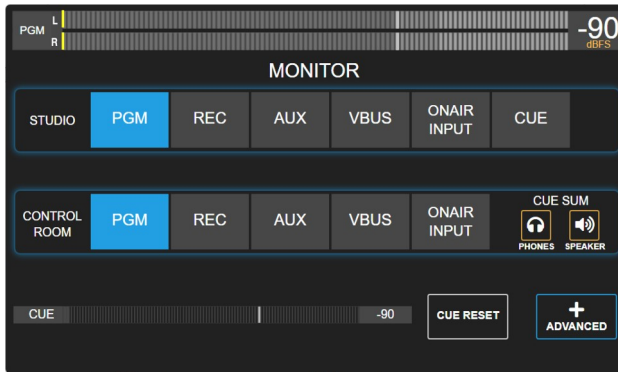


Image 33: Monitoring switcher

**STUDIO:** Allows to select a signal source for the Studio Monitor outputs. The options are:

PGM, REC, AUX, VBUS, ONAIR INPUT, CUE

The send level for speakers and headphones of the Studio can be adjusted in the MONITOR section (see below). Usually this level is fixed and each user within the studio adjusts their level from a headphone distributor.

For radios operating in a single room, the monitor speakers can be connected to the Control Room output which has a physical volume knob.

**CONTROL ROOM:** Allows to select the signal source for the Studio speakers and headphones. The options are:

PGM, REC, AUX, VBUS, ONAIR INPUT

### MUTING THE CONTROL ROOM SPEAKERS

When a microphone is used into the Control Room, it can be configured to mute the CR speaker output when is turned on (see 5.6 The screen 'Channel' - Item 8).

The above is also used in radios that works in a single-room, with no separation between the Studio and the Control Room. In this case, the monitor speakers can be connected to the CR speaker output, in order to use the volume knob on the console panel (note that there is not volume physical knob for Studio Speakers).

In this case, all the microphones are configured as located into the Control Room, to activate the muting for the Control Room speakers.

The options **CUE TO SPEAKER** and **CUE TO HEADPHONES** inject the CUE signal into the speakers and/or headphones of the Control Room.

When these buttons are activated, the CUE signal mixes with the currently monitored signal. **The monitored signal reduces its level automatically when a CUE signal is present.** The degree of attenuation can be defined in the MONITOR → ADVANCED section.

The CUE signal can be assigned to an output, for connection of **auxiliary CUE loudspeakers**. By default, CUE is present on OUTPUT 4.

### CUE METER

Shows the CUE level in dBfs. Is the level **after the CUE level knob**; therefore, there is no indication if the knob is closed.

### CUE Reset

The CUE Reset option turns off all active CUE buttons, both physical buttons and virtualized.

### 5.4.1.1 MONITOR → ADVANCED

This screen allows the user adjust the level of each monitoring signal. Also contains some configuration options.

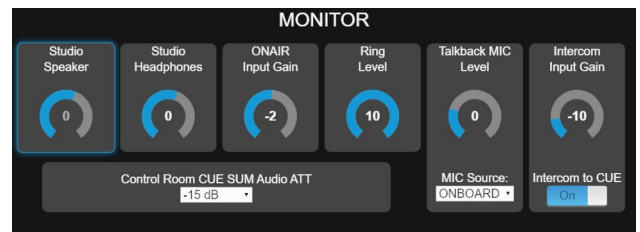


Image 34: monitoring settings

### Navigation



Turn the encoder to navigate the controls.



Push the encoder to select a control.



Turn the encoder to change the value of the selected control.



Push the encoder to confirm the new value. When do it, the encoder returns to the navigation mode.



The user can directly touch a control to select it, and then turn the encoder to change the value. To edit the settings, directly touch on the item.

### STUDIO SPEAKER

Adjust the level for the "Studio Monitors" (on the RJ45 "Studio Monitor" located on the rear panel). This is a line-level output specifically designed for use with active speakers.

### STUDIO HEADPHONES

Adjust the level for the "Studio Headphones" (on the RJ45 "Studio Monitor" located on the rear panel). This is a line-level output that allows to connect external headphones distributors.

### NOTE: MONITORING OUTPUT LEVEL

To change the level for studio loudspeakers and studio headphones, take in mind that a very low level at the console's outputs will need too much gain in the loudspeaker and headphones amplifiers, causing SNR loss.

## ON-AIR INPUT GAIN

Gain control for the input "On-Air Monitor Input" (external tuner). This input can be routed to a fader, to be used as a standard line input.

## RING LEVEL

Set the volume for the "ring tone" that sounds in CUE with the incoming calls (for POTS and cell phones).

## TALCKBACK MIC LEVEL

Set the level for the talk-back mic signal, which is send to the Studio monitoring and TELCO.

## TALKBACK MIC SOURCE

Set the microphone used for talk-back (studio monitoring and TELCO).

The options are:

- The built-in microphone
- The inputs MIC-1 to 5; UNIDEX-4\_L; UNIDEX-4\_R; DANTE 1 & 2.

A MIC input that was assigned as talk-back, can be assigned to any channel at the same time. This enables the operator to use this mic as private talk-back or to talk on-the-air.

## CONTROL ROOM CUE SUM AUDIO ATT

Set the attenuation value applied to the monitored signal at the Control Room when the CUE signal is mixed (on speakers or headphones). This attenuation value is also applied when the talk-back mic is activated.

### 5.4.2 Control room, Master-MIC, talk-back



Image 35: Monitor

**1 MASTER ON-AIR MICS:** Turns on/off the microphones and groups assigned to MASTER MIC.

**3 CONTROL ROOM MONITOR LEVEL:** Allows to adjust the level of the speakers connected to the Control Room output. This output is designed for connecting active or powered speakers. The maximum level attained will be affected by the volume adjustment specific to the monitors.

**4 HEADPHONES LEVEL:** Manages the level of the operator's headphones (connected to Control Room Headphones).

**5 CUE LEVEL:** It controls the level of the CUE signal. The main screen features a CUE level meter, in dBfs. This meter shows the level after the knob "CUE"; therefore, there will be no indication when the knob position is zero.

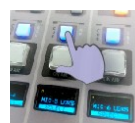
## 5.5 The function SELECT

The **SELECT mode turns the CUE buttons into channel selection buttons**. It is activated when accessing the CHANNEL and PROCESS screens. When is active CUE buttons lights in blue and the OLED indicates "SELECT." SELECT mode allows to choose a channel to view their parameters or perform actions on it.

Three actions can activate the **mode SELECT**:

1. Choosing CHANNEL at the main screen.
2. Choosing PROCESS at the main screen.
3. Choosing SELECT on the CHANNEL screen or a PROCESS screen (EQ, compressor, etc.).

Once SELECT is active, it is possible to **choose a channel** in two ways:



### METHOD A

Pressing CUE/SELECT selects the channel. The SELECT mode remains active, allowing the selection of other channels (facilitating easy comparison/review of channels). The CUE/SELECT button on the selected channel continues to flash.



### METHOD B

Turning the encoder to navigate the channels. The encoder navigates the 18 channels and the four groups.



To confirm the selection, press the SELECT button or push the encoder. Upon confirmation, the SELECT mode is deactivated, and the CUE/SELECT buttons return to their normal operating mode (CUE). SELECT is also deactivated if a control on the screen is touched.

## 5.6 THE SCREEN 'CHANNEL'

The screen 'CHANNEL' displays the status values and settings of the selected channel. From this screen the operator accesses basic functions, such as the main fader or the input gain. Access to other advanced settings requires access as ADMIN profile.

### NOTE

The UX18 has 3 users profiles. Access to advanced configuration options is only allowed for the ADMIN.

To access the CHANNEL SCREEN proceed:

1. **Tap on the 'CHANNEL' option** on the main screen. It will display the last selected channel and activate the "SELECT" mode. Notice that the menu option "CHANNEL" changes to "SELECT".

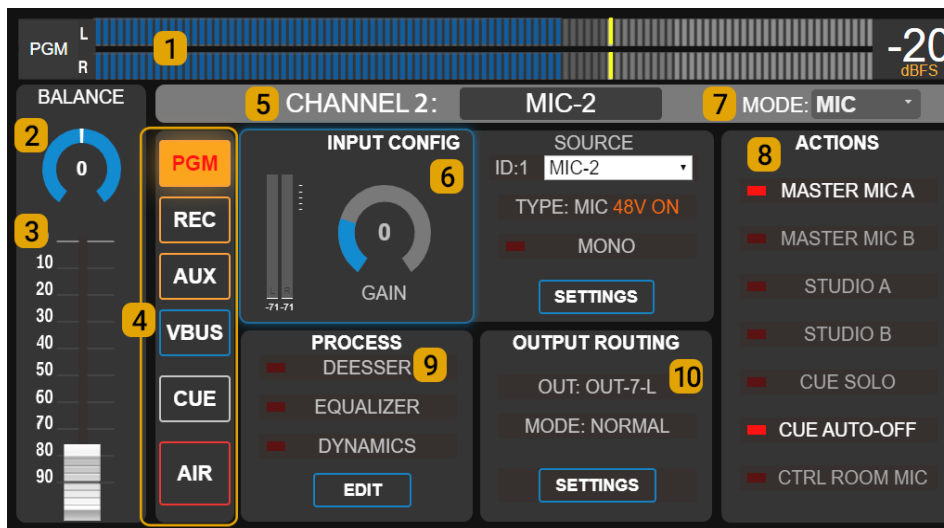
2. **Choose a channel.** The hardware buttons CUE/SELECT allows selecting a channel. While the SELECT function remains active, different channels can be successively selected to compare their values.

VIRTUAL CHANNELS selects by turning the encoder, which navigates the 18 channels and the 4 groups. The channel's name zone changes its color for those channels that are assigned to a group.

3. To confirm the channel and exit the SELECT mode, tap on the SELECT option on the screen or push the encoder. The select mode is also canceled when a parameter is edited for the selected channel.



## 5.6 THE SCREEN 'CHANNEL'



**Channel's screen.** Displays config status and the basic controls for a selected channel. Allows to change the input/output routing and the audio processing (according to the user profile permissions).

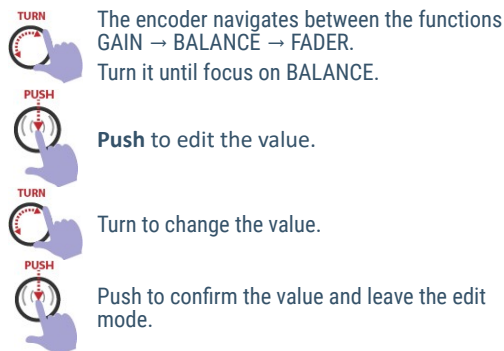
### 1 VUMETER

It shows the audio level for the selected BUS. Tapping on the label 'PGM' (default) switches the meter to AUX, REC, and VBUS successively.

The value displayed to the right of the meter shows the peak value in VU or dBFS, with the peak retained. Tapping on the value changes the unit. This setting will take effect across all screens.

### 2 BALANCE

In stereo channels it adjust the balance between Left and Right. In mono inputs is the PAN-POT control. To change the value:



### 3 MAIN FADER

The on-screen fader is a virtual version of a hardware fader and it works in the same way: to set the mix level for the signal to be mixed on all assigned buses. For hardware channels, the on-screen fader reflects the current position of the mechanical fader.

If the user moves the on-screen fader, the signal level will change, but the mechanical fader remains in the same place, as they are not motorized.

In this condition, the OLED screen in the channel will show the level value flashing, indicating that it doesn't match the position of the hardware fader

### NOTES

Although virtual channels are 100% operational from the touch screen, for convenient operation it is recommended to use the remote web interface, which can run on a larger screen or tablet.

**The level at the outputs and VU-meters indication can differ since outputs have gain offset.** Level meters and outputs matches when output gain control is set at the nominal level established for this output (nominal levels are indicated on screen).

### 4 BUSES

For **hardware channels**, it reflects the status of the PGM, REC and AUX buttons. The hardware buttons can be managed from the touch screen (and from the remote web interface).

For **virtual channels**, they allows assigning a channel to the mix buses. The V-BUS is not on the hardware and can only be assigned from the screens CHANNEL and VIRTUAL.

### 5 NOMBRE

Allows entering a name for the channel. The maximum extension is 10 characters. This name will appear on the OLED display.

### 6 INPUT CONFIGURATIONS

#### NOTE

The UX18 has three users profiles. Access to advanced configuration options is only allowed for the ADMIN.

It contains the input gain control and shows information relative to the input associated with the channel. Access to SETTINGS and input assignment is restricted to the user profile ADMIN.

## INPUT GAIN

To set the input gain (dB), proceed:



The encoder navigates the options GAIN → BALANCE → FADER.

**Turn the encoder** until focus on GAIN or touch the GAIN control.



**Push the encoder** to edit the value.



**Turn the encoder** to change the value.



**Push the encoder** to confirm the value and exit.

## 7 MODE

The mode changes the behavior and some options available on the channel. It is set according to the type of signal that the channel will manage. To change it, touch the option on the screen and select a mode from the drop-down menu. The channel can be set in three modes:

**LINE** This mode is for channels that manage any type of source other than microphones or communications

**MIC** It is the mode used for channels that handle microphone signals. Microphones can be connected to the rear-panel mic inputs, external UNIDEX adapters (UDX-2MIC and DMIC), wireless microphone receivers, or third-party preamps connected through line inputs.

**TELCO** It is the mode used for channels that manage communications (land lines, VoIP, Bluetooth cell phones). Channels defined as TELCO can remotely control external adapters like UDX 2TEL, UDX 2TEL-P, and UDX TEL+BT

## 8 CHANNEL ACTIONS

This option defines the behavior of the channel and presents some special features. The available options here will change according to the channel's MODE.



Each action is enabled or disabled on the CHANNEL SCREEN, by touching the option on the screen. The red indicator shows if the option is enabled (on) or disabled (off).

## CUE SOLO and CUE AUTO OFF

This features are available on all channels.

### CUE SOLO

If enabled, when a CUE is turned on, all CUEs of other channels set as CUE SOLO turn off. Only the channels working as 'CUE SOLO' are affected. For other channels, one or more CUE buttons can remain on.

If all channels work in CUE SOLO mode, there will always be only one CUE turned on.

A usual configuration is to work the lines in 'CUE SOLO' mode (computers and audio players) and the microphones and TELCO channels in 'CUE NORMAL' mode.

### CUE AUTO OFF

If enabled, CUE is turned off when the channel is on-air (the ON-AIR button is activated and the fader raised).

## ACTIONS FOR "MIC" CHANNELS

The following actions are available only when MODE=MIC.

### MASTER MIC

Allows the user to assign the channel to the master-mic button. All channels assigned to the master button are turned on simultaneously when the button is pressed.

A channel assigned to a master button can always be activated independently by pressing the ON-AIR button on the channel.

**The groups channels can be assigned to the MASTER-MIC**, in order to work whit grouped microphones.

### CTRL ROOM MIC

This option must be enabled when a microphone is located in the Control Room. It causes the Control speakers to be muted when that microphone is turned on, but not the Studio speakers

## 9 PROCESS

It shows the available audio processing for that channel, and whether they are activated. Each processing can be turned on/off by touching the process name (EXPERT and ADMIN profiles).

To change the settings, touch the EDIT option (only for ADMIN profile). The available processes are not specific to the channel but depend on the input assigned to the channel

## 10 OUTPUT ROUTING

It shows if the channel is associated with a direct output. An example of this is the external adapters UNIDEX UDX 2TEL, which have an associated output for sending signals to telephone lines.

The ADMININ user can access the configuration by touching SETTINGS..

## 5.7 TELCO - COMMUNICATIONS

This screen is launched by pressing the button "TELCO" (located at the right of the screen).

The "TELCO" screen only shows the channels with MODE=TELCO which are those associated with land telephone lines (POTS) and Bluetooth mobile phones.

One cell phone can connect to the built-in Bluetooth. Land lines and additional cell phones connect to UNIDEX using external adapters.

In the following explanations, the term 'TELCO line' refers to both traditional telephone lines and mobile phones.

Calls made on traditional telephone lines and mobile phones connected via Bluetooth are operated in the same way on the console, with some differences based on the specific circumstances of each case (for example, when making the call)."

**Each line uses its own channel.** Any channel can be configured in TELCO mode, but hardware channels should preferably be used.

The main operations (pick up, hold, send on air, hang up) can also be done using the CUE and ON AIR buttons on each channel (see below).



### 5.7.1 THE SCREEN "TELCO"

**1 LINES:** UX24 supports up to 8 TELCO lines. Two groups of four channels are displayed in this area. When there are more than four lines, the "Page Down" button is enabled to access the rest.

Each row corresponds to a line and indicates:

- The number and name of the channel.
- The status of the line: YELLOW if there is an incoming call (RING) GREEN when a line is on hold and RED when it is on the air.
- Time on-air or time on-hold.

**2 SELECTED LINE:** It shows the channel number and the name of the currently selected line. The zone at right shows the controls for selected line.

To select a line, touch it on the list. Regardless of the line status (ON AIR in red, ON HOLD in green, RINGING in yellow, or INACTIVE in gray) the right zone shows the controls. When a line is selected, it is highlighted with a blue frame and an arrow at the right.

**3 INPUT LEVEL:** Shows the audio level of the incoming signal. It also indicates the name of the UNIDEX input to which the UDX adapter was connected.

**4 INPUT GAIN:** Display the input gain control that corresponds to the input assigned to the UDX adapter. Use the encoder to change the value.

**5 SEND LEVEL:** Displays the output gain control that correspond to the output assigned to the UDX adapter, for sending audio to the line. Use the encoder to change the value

**6 TALK:** Activates Talkback mode, enabling the operator's microphone to speak to the line. Equivalent to holding down the CUE button of the channel, but unlike CUE, the TALK option on the screen does not operate with the logic of a push button, but rather as a switch: it is activated with one tap and deactivated with a second tap.

It also deactivates when the line is put on air. While TALK is active, either by pressing CUE or by turning it on on the screen, the transmission of the PGM signal to the line is interrupted, only the talk-back microphone is sent.

**7 DIAL:** Launch a DTMF touch keyboard to dial from the console (only for POTS).

**8 HOLD/HANG UP:** This icon has a double function:

- With incoming calls, it allows to take the line and leave the call on HOLD.
- When a call is on-air, it allows to end the communication (HANG UP)

**9 CONFIG:** Reports the type of adapter associated with the selected line, and its settings options. The options displayed changes according to the UNIDEX adapter, and are the following:

#### NULL ADJUST

This option is shown only for POTS lines (UDX 2TEL; 2TEL-P and TEL+BL). Allows to adjust the rejection factor from a preset located in the external device (see 3.3 - UDX 2TEL and 2TEL-P).

#### BLUETOOTH PAIRING

This option is shown for TELCO Bluetooth channels (internal and external adapters UDX TEL+BL and UDX 2BLT). It turn on the Bluetooth pairing mode to connect a cell phone.

## CONNECTING UDX EXTERNAL ADAPTERS

To know how to install the external adapters UDX 2TEL, UDX TEL+BL and UDX 2BLT see Section 3 – UNIDEX External Adapters.

Installation guides for each UDX device can be downloaded from the web site. Please see Section 7 – Diagrams and Tech notes.

## 5.7.2 Using TELCO with Mobile Phones

### 5.7.2.1 Pairing the mobile phone

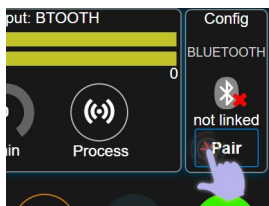
When linking the mobile phone to the console, a wireless connection is established between both devices. The phone records the unique Bluetooth identification code of the console. Connected to the Bluetooth mobile, the console broadcasts all types of calls made by the mobile phone (via telephone network or generated by apps).

## ABOUT THE CELL PHONE LOCATION

While the cell phone can be up to 5 meters away from the UDX Bluetooth adapter, it is recommended to keep it close to avoid interference in communication, which can result in audio artifacts (the "robotic" effect).

The procedure for pairing a phone for the first time is as follows:

1. On TELCO screen, select with a touch the Bluetooth device that you want to pair.
2. Press the BLUETOOTH PAIRING option.



Bluetooth on the UDX adapter will be activated in pairing mode. The OLED screen on the channel will indicate the "pairing" status.

3. Turn on Bluetooth on the phone and search for devices. SOLIDYNE BTM641 will appear in the list "found devices".
4. Connect to SOLIDYNE BTM641. The phone is paired and connected to the channel's Bluetooth.
5. The TELCO screen and the channel's OLED now indicate PAIRED.

## BLUETOOTH AUDIO LEVEL

On the mobile phone, turn up the Bluetooth volume to the maximum to achieve the best signal-to-noise ratio.

The adjustment must be done with an active Bluetooth call. The phone volume is independent of the Bluetooth device volume. If the volume is adjusted without establishing a Bluetooth call, the volume of the phone's internal speaker changes, not the volume of Bluetooth.

Note that this is a phone setting. Check the volume every time you connect the phone to the console.

### 5.7.2.2 Bluetooth disconnection and re-connection

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

#### Turning Bluetooth on/off on the mobile device:

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

#### Pairing the mobile device with a Bluetooth device:

This refers to the mobile phone finding and remembering an external Bluetooth device (such as headphones, the car's audio center, or a UNIDEX channel). This procedure is only done once for each device. Once paired, the device can be unpaired, which means that the phone "forgets" it, and the pairing procedure will need to be repeated to re-connect it.

#### Connect/disconnect a Bluetooth device to the mobile:

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

When Bluetooth is enabled on the mobile phone, 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).

As with any conventional Bluetooth device, the mobile phone will "disconnect" from the console in the following cases:

- **By turning off Bluetooth on the cell phone.** Generally, this action does not end an ongoing call; it transfers it to the phone. But this may vary on each phone. Reconnection is re-established by turning on Bluetooth on the phone. In some cases, reconnection is automatic. In others, it will be necessary to select the SOLIDYNE BTM641 device again from a list. If the call was not interrupted, it will be available again on the console upon reconnection.
- During an ongoing call, when **the phone is disconnected** from the Bluetooth device/channel. 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, reconnection will not be possible if the same Bluetooth channel on the console was paired with another phone and it is now connected.

### 5.7.2.3 Linking more than one Bluetooth phone

Up to 8 TELCO lines can be used linked to mobile phones (using UNIDEX adapters). If there are two or more UNIDEX Bluetooth interfaces, the procedure for linking a second phone is the one already described in 5.7.2.1, but taking into account some particularities.

Let's say there are three Bluetooth links available at the mixer, named (CHANNEL NAME) as BLT 1, BLT 2, and BLT 3. BLT-1 has already been linked to "Mobile A" and now the

user wants to link a second "Mobile B" to BLT-2. The procedure is as follows:

1. Mobile-A, linked to BLT-1 channel, can be connected or disconnected from the console. It doesn't matter.
2. On the TELCO screen, select the BLT-2 line with a touch and activate the Bluetooth PAIRING option that appears on the screen.
3. On Mobile-B, activate Bluetooth. When doing so, the phone performs a search and displays a list of found devices. This list only shows devices visible in pairing mode (waiting for connection). SOLIDYNE BTM641 will appear there. This will be the device corresponding to the BLT-2 channel, since it is the only UNIDEX device in PAIRING mode.
4. Connect Mobile-B to SOLIDYNE BTM641. The second phone is now linked and connected to the BLT-2 channel of the console.

**IMPORTANT NOTE:** All UNIDEX Bluetooth devices have the same name: SOLIDYNE BTM641. This means that the different Bluetooth channels (regardless of the name assigned to each channel) on the phones appear as SOLIDYNE BTM641. This is not a technical problem, because each Bluetooth line is paired to a different phone, so when a connection is re-established on a phone, it will always connect to the line to which it was paired.

However, we recommend editing the name of the SOLIDYNE BTM641 device after pairing each phone, so that it matches the name assigned to the channel on the console. Following the previous example, the SOLIDYNE BTM641 device on Mobile-A will be renamed as UNIDEX BLT-1, on Mobile-B it will be UNIDEX BLT-2, and so on.

This makes it easier to recognize which channel each phone is connecting to. For example, in the case of a "external" smartphone, such as a producer or journalist's personal mobile, which has been linked to two or more Bluetooth channels at once, to connect to any of them in case some are already busy by other phones from the radio. By identifying the devices, the operator can indicate which channel to connect to, and the person can quickly locate it on their phone.

#### BLUETOOTH DEVICE'S NAME

On the cell phone, the user can always edit the default name "SOLIDYNE BTM641" by accessing the Bluetooth device options. This feature is usually simple and varies depending on the mobile operating system version.

#### 5.7.2.4 Incoming call on mobile phone

1. The mobile phone that receives the call is connected to the console via Bluetooth.
2. When an incoming call is received, the channel's CUE/SELECT button will flash blue, and at the same time, the ring tone is heard on the CUE circuit. The TELCO display line indicates RING.

In the previous condition, the call can be answered in two ways:

- a) By pressing CUE/SELECT on the channel.

- b) Touching HOLD on the TELCO screen.

3. When answering, the call remains "on hold". The caller is not on the air, but listens to the program signal. This condition is indicated on the screen (HOLD in green) and by the ON AIR button that keeps flashing.

#### CALLS FROM APPs

When calls are originated using applications (such as WhatsApp, Skype, Hangouts, among others), many mobile phones prior to 2020 DO NOT support answering via Bluetooth. In this case:

- a. Answer the call directly on the phone.
- b. Then press ON AIR on the corresponding channel.
- c. The call will be put on HOLD on the console.
- d. Proceed with the normal operation (see Point 4).

4. To speak on the private circuit, press and hold the CUE button on the channel. While the button is held down, the talkback microphone is exclusively sent to that line, the OLED display shows TLBK, and the monitoring in the Control Room is silenced.

When the CUE button is released, it returns to the previous state (the talkback function does not change the state of CUE). With CUE turned on, we can hear who is on the other end of the line through the CUE circuit.

5. To put the call on the air, press the ON AIR button on the channel and raise the fader.
6. To put the call on hold again, press ON AIR.

Remember that the mobile phone can resume the call by disconnecting the Bluetooth device; and return it to the console by reconnecting Bluetooth.

7. To end the communication:

- From the channel: hold down ON AIR for 3 seconds.
- From the TELCO screen: touch HANG-UP.

#### IF BLUETOOTH ON CELL PHONE IS DISCONNECTED

If the mobile phone is not connected to UNIDEX at the time of receiving the call, answer the call and then activate/connect Bluetooth on the mobile phone.

Once the mobile phone is connected to a Bluetooth channel, press ON AIR on the channel to put the call on hold.

#### 5.7.2.5 Making the call from the cellphone

When making the call from the cellphone that is linked to the console, the Bluetooth status on the cellphone must be taken into account. There may be two situations:

If Bluetooth is enabled and connected to an UNIDEX channel:

- Make the call from the mobile phone. It can be a conventional call (to a phone number) or a voice call over any installed application (WhatsApp, Instagram, Discord, etc.)

- On the console channel, press ON AIR to transfer the audio of the call (it will be on hold).
- When the contact answers, talk from the console using the talk-back microphone and the CUE circuit.

**If Bluetooth is disabled or NOT connected to a UNIDEX channel on the phone:**

- Make the call, speak and then activate/connect Bluetooth on the mobile. If Bluetooth was disabled, it will usually reconnect to the console when activated. If Bluetooth is enabled on the phone but not connected, connect it by selecting the UNIDEX device from the list of Bluetooth devices on the phone.
- Bluetooth can also be enabled before making the call and once connected to the console, dial.
- On the console channel, press ON AIR to transfer the audio of the call, which will be on hold.
- Talk from the console using the talk-back microphone and the CUE circuit.

### 5.7.3 Communications generated by external software

In recent years, voice communications over the Internet have become increasingly popular, either as "voice calls," an alternative to traditional phone calls (such as WhatsApp, Discord, and others), or various video-integrated meeting software such as Zoom or Google Meet.

When these communications are generated from a Bluetooth-linked phone, they are handled as TELCO lines.

When they are generated from an **external computer**, they cannot be configured as TELCO lines. Instead, they are managed as a line-level audio signal, but with some particularities:

- The **UNIDEX USB-OUT** that is assigned as the input device in communication software is configured to send PGM signal in mix-minus mode. This ensures that the communication software never receives its own audio, which generates cleaner communication in terms of artifacts and audio fluctuations.
- In the USB playback channel, the operator can use the CUE button in talk-back mode (by holding it down) to privately talk with the PC software, just like how it operates with TELCO lines.
- The communication in the computer can be in conference with one or more TELCO lines.

### 5.7.4 Using land lines (POTS)

#### 5.7.4.1 Incoming call

1. When a call is received, the channel's CUE button will flash in blue and a ring tone will be heard on CUE. The TELCO screen also indicates RING on the corresponding line.
2. The call can be answered in two ways:
  - a) Pressing the CUE/SELECT button on the channel.

- b) Touching HOLD on the TELCO display.

When answered, the line is taken and the call is on HOLD mode. The line's audio does not go on-air, but the caller hears the program signal.

3. To talk privately with the line, press and hold CUE on the channel.

While the button is pressed, the talk-back microphone is sent to that line, replacing the program signal, and the Control Room monitoring output is muted. The OLED indicates the TLBK mode.

When CUE is released, the CUE/SELECT button returns to its previous state (the talk-back function does not change the state of CUE). When CUE is active, the line's audio is heard through the CUE circuit.

4. To put the call on the air, press the ON AIR button on the channel and raise the fader.
5. To return the line to "HOLD", press ON AIR again.
6. To end the call:
  - Hold down ON AIR for 3 seconds.
  - Touch HANG-UP on the TELCO display.

#### 5.7.4.2 Making a call from the console

The DIAL keypad only works with the landlines. To make a call with a Bluetooth line, it must be done from the mobile phone. To dial, proceed as follows:

1. Go to TELCO screen.
2. Select a landline (not a Bluetooth).
3. Tap the DIAL icon. A keypad will launch to dial the number. This action puts the line on HOLD mode. The ring tone is heard on the CUE circuit.
4. Dial the number. The numbers are dialed directly as they are entered.
5. When the call is answered, a private conversation is held on the CUE circuit.
6. From here, proceed with the operation as indicated in 5.7.4.1 Incoming calls (from Point 10).

#### AUDIO LEVELS ON AIR AND LANDLINES

Do not exceed the level of the telephone line on air. Telephone hybrids have a high rejection but not absolute. The audio sent to the line (the voices from the studio) is not 100% cancelled and returns through the hybrid, adding to the studio voices (this is what optimizes the rejection adjustment). With normal work levels, this effect is not noticeable; but if the hybrid is worked at high levels, the studio voices can become "dirty" (coloration).

The calling may not reach 0VU to match the loudness of the studio voices peaking at this same level. The lows carries the greatest energy load, causing greater deviation in the indicators. Lows are dumped in the audio of the calls.

For the ear, loudness is defined by the range of middle frequencies. If you notice that the VU meter "peaks lower" with a telephone line compared to the studio voices, this does not mean that "it sounds lower". In addition, the audio processor of the transmission will match both signals.

### 5.7.4.3 Answer/Make remote calls

For setups where calls are answered by a production team or phone operators, the UDX-Producer adapter allows for transferring calls to UNIDEX from a remote location. Calls are directly put on hold mode. During times when calls must be handled from the console, the UDX-Producer adapter is switched to passive mode, and calls enter directly into TELCO on UX18

## 5.8 AUDIO PROCESSORS

The audio processors are implemented fixed on each of the inputs. They are not "on the channel" and cannot be reassigned between channels, buses, or outputs. The processes enabled on a channel depend on the type of input assigned to that channel.

INPUT	PROCESS		
	EQ + Filters	Comp.+Gate	De-esser
MIC	3 bands	si	si
UNIDEX	3 bands	si	no
USB	3 bands	no	no
AES3	3 bands	no	no
BALANCED	3 bands	no	no

To access to the audio processors, press the button PROCESS at the main screen. This launch the process screen and enables the function SELECT to choose a channel (for details please see 5.5 – The function SELECT).

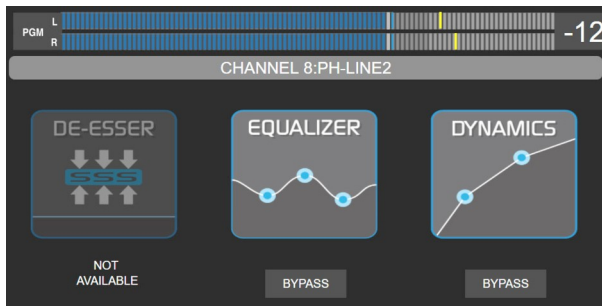


Image 36: Processing screen

The screen PROCESS allows:

- Know which processors are available for the selected channel; and if they are active or bypassed.
- Enable/disable each processor by touching the BYPASS option.
- Access the settings screen of a processor, by touching the on-screen icon, or the associated contextual button.

#### SELECT A CHANNEL:

- When the function SELECT is active, the buttons CUE/SELECT on the surface allows to select a channel.
- Turning the encoder navigates the 18 channels (hardware and virtual). On screen the user see the processors status for each channel.

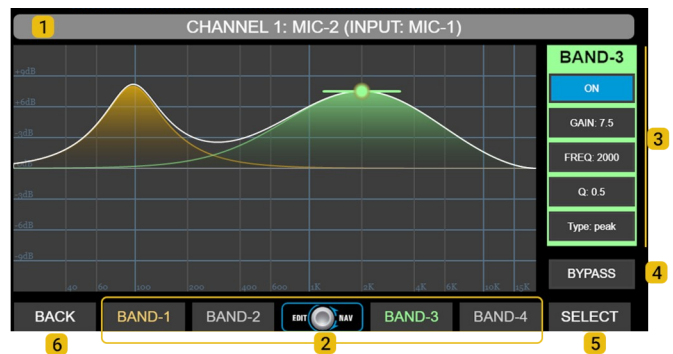
- For the GROUPS channels, the PROCESS screen will show the status of the process of each channels assigned to this group.

#### NOTA

If the SELECT mode is active, any action on the PROCESS screen cancels it. The action will take effect on the currently selected channel at that time.

"SELECT" can be reactivated from the process screen to switch to another channel

The parameters within each processor are accessed using the encoder and through touch options. In the following explanations, it is detailed for each case which parameters are controlled by the encoder and which are



touch-sensitive.

### 5.8.1 PARAMETRIC EQUALIZER

#### Navegación

Girar para elegir una opción. El encoder navega en bucle entre las funciones → GANANCIA → FRECUENCIA → Q



Las funciones ON, TYPE y BY-PASS se cambian tocando directamente la pantalla.

Las curvas de ecualización se pueden cambiar tocando la pantalla, modificando su frecuencia, la ganancia y el Q, que se cambia variando el ancho con dos dedos.

**1 NAME BAR:** Shows the channel # and the label. When a channel is assigned to a GROUP, the NAME zone will change its color according to the assigned group.

**2 EQ BANDS:** Band selection. The control zone at the right shows the values for the selected band. Each band identifies with a color.

**3 CONTROLS.** Shows the parameters of an EQ band. The background color changes with the selected band. Equalizer settings can be saved in the current scene.

**ON/OFF:** Turn on/off the band. To switch the status, touch the option on screen.

**GAIN:** Set the gain in a range of  $\pm 9$  dB.

**FREQ:** Set the center frequency. Can work from 20 Hz to 20 KHz.

**Q:** EQ wide.

**TYPE:** EQ mode. To change between the options touch the item TYPE on screen. Band 1 can work as *peak, low-shelf or hi-pass filter (HPF)*. Bands 2 and 3 can be *peak; HPF or LPF*. Band 4 can work as *peak; hi-shelf or low-pass filter (LPF)*.

**4 BY-PASS:** Disables the EQ. Bypass also appears into the PROCESS screen. To switch its status, touch the icon on screen.

### 5.8.1.1 LOW-PASS/HI-PASS FILTERS

The Band #1 can work as a 2<sup>nd</sup> order Hi-Pass Filter (12 dB/oct).

The Band #3 can work as a 2<sup>nd</sup> order Low-Pass Filter (12 dB/oct).

Band #2 can work a HPF or LPF. Setting two bands with HPF/LPF at the same frequency, the attenuation slope will be 24 dB/oct.

### 5.8.2 DYNAMICS



Image 37: Dynamic compressor

#### Navigation



Turn to choose an option:  
→ GATE THRESHOLD → COMP THRESHOLD → RATIO → OUT GAIN → ATTACK → HOLD → RELEASE →



The functions ATTENUATION, STYLE, BY-PASS are change using the touch screen.

The DYNAMIC stage has a noise gate and a wideband dynamic compressor. Processor settings can be saved in the current scene.

**1 Transfer curve:** Graphical representation of the dynamic processor input/output curve, including the gate and the compressor.

**2 LEVEL:** Peak level at the compressor input (dBfs)

**3 GATE:** Noise gate.

**GATE Threshold:** Level below which the gate closes.

**ATTENUATION:** Gain reduction applied to the signal when the gate acts.

**STYLE:** This option changes the recovery time between preset values.

**ACTIVE:** This indication lights red when the gate closes.

**4 COMPRESSOR:** Standard controls for a compressor: threshold, ratio, attack and recovery times and reduction meter.

**5 BYPASS:** Disable the Dynamic processing.

### 5.8.3 DE-ESSER

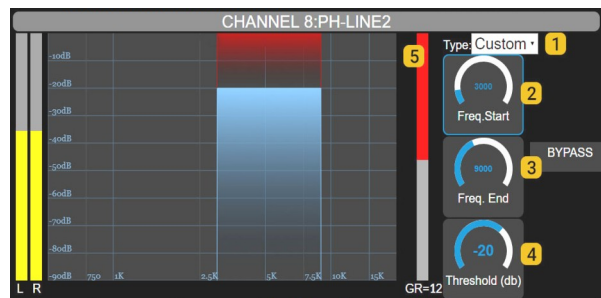


Image 38: De-esser

#### Navigation



The encoder navigates the options:  
→ FREQ.START → FREQ.END → THRESHOLD →  
To edit proceed as in the previous cases.



The features TYPE and BY-PASS are switched using the touch screen.

MIC-1 to MIC-5 inputs feature a de-esser for reducing sibilance. The de-esser operates like a multiband compressor, attenuating only the frequency range set when the signal level exceeds the threshold value.

It features the following controls:

**1 Type:** Presets. Changes the behavior of de-esser.

**2 Frequency start:** Lower frequency for the working range. Usual values are between 4.5 to 5.5 KHz, according to the characteristics of the voice.

**3 Frequency end:** Upper frequency for the working range. Usual values are between 7 to 10 KHz, according to the characteristics of the voice.

**4 Threshold:** Level in dB. When the signal level within the working range falls below the threshold, it is attenuated in a 15: 1 ratio

**5 Gain reduction:** Shows the attenuation in dB.



## 5.9 Scenes and user profiles

Access is done by pressing the SCENES option on the main screen (HOME).

**Scenes** allow for quick switching between 12 stored configurations, defining the assignment of inputs to channels, channel groups, signals assigned to outputs, gains, and audio processor settings.

**User profiles** allow restricting access to certain console functions.

The screen for user and scene management allows changing the active user, changing the current scene, and contains advanced functions for swapping channel positions or reorganizing them.

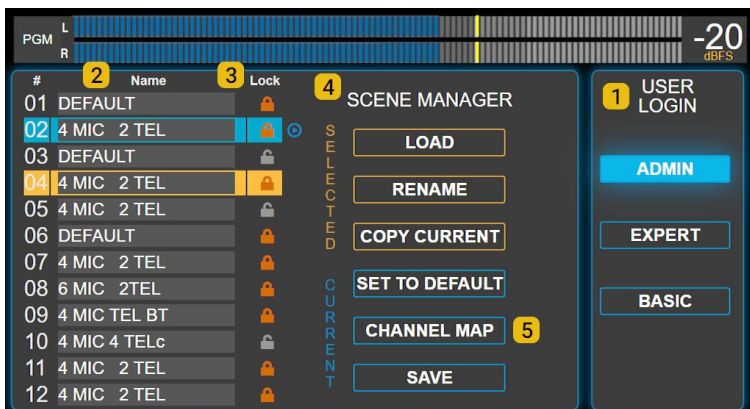


Image 39: Scenes and users

### Navigation



The encoder navigates the list of scenes. Push to load the selected scene.



All options operates using the touch screen.

### 5.9.1 User profiles

**1 USER LOGIN:** This area of the screen displays the currently active user (logged) and allows switching to another user by tapping the option on the screen

There are three user profiles with different access permissions. The user profiles for the on-board screen and the remote web interface are the same, but each interface have their own login.

#### BASIC USER

Is the profile with more restrictions. Logged with this user, the operator:

- Is the profile with more restrictions. Logged with this user, the operator:
- Don't needs password to login.
- Can't access to advanced settings.

- Can load scenes, but can't save changes neither create new scenes.
- Can see the settings and status of the processors, but cannot modify their parameters or turn on/off processors in the scene.
- Can temporarily change basic channel options: GAIN; LABEL; PANNING; CHANNEL ACTIONS. Is not allowed to save changes to the scene.

#### EXPERT USER

This profile has a medium level of restriction. Logged with this user the operator:

- Needs the password to login (default: expert). The password can be changed only by the ADMIN.
- Don't have access to advanced settings.
- Has full access to the processors.
- Can change the input gain, panning, label, input assignment and channel mode using the CHANNEL screen.
- Can save changes to current scene, except for the protected scenes (which requires ADMIN profile).
- Can create new scenes, even starting from a protected scene.

#### ADMIN USER

- It has password protection (default: admin). Allows you to change passwords from the SETTINGS screen.
- Can create protected scenes (read only) that can be used with all profiles but only can be changed by the ADMIN profile.

When booting, UX18 loads a predefined user profile (by default is ADMIN). **On startup, the password is never requested.**

The boot user and passwords are set from SETTINGS → USERS (see 4.9 SETTINGS → USERS).

### 5.9.2 The scenes

A scene stores all the console settings, except for system settings (network, streaming, startup user, etc).

Input, output, channel, gain, bus assignment, and audio processor settings are **saved in the scene.**

Scenes allow for quickly loading a predefined configuration according to the requirements of each show; for example, changing the inputs assigned to certain channels, the channel order, or microphone processor settings.

#### TAKE IN MIND

To permanently store a change, the scene must be saved/updated. For example, every time a parameter is modified, such as a gain, an equalizer, or a channel label.

**2 SCENES:** This area (see previous image) displays a list of 12 scenes. To load a scene proceed:

- Turn the encoder to select and press it to confirm.
- Touch the scene in the list and then "LOAD".

The currently active scene is indicated in the list highlighted in light blue. The selected scene is highlighted in orange. All scenes can be modified by the user, except for protected scenes that can only be modified by ADMIN

**3 LOCK:** The ADMIN user profile can create protected scenes (LOCKED): The scenes are indicated with a lock icon. EXPERT and BASIC profiles can load protected scenes, but cannot modify or delete them.

**4 SCENE MANAGER:** Shows two groups of actions: for the selected scene and for the current scene.

**LOAD (SCENE):** Load the selected scene. Loading can be done by pressing the option on the screen; or by turning and pressing the encoder.

When loading a scene, a confirmation message will appear. If the action is not confirmed, the action is automatically canceled after 8 seconds. All user profiles can load any scene.

A scene can be loaded with one or more channels actives on the air.

- Channels remain on-air even if they change position, maintaining bus assignment.
- If the new channel position has a different attenuation value, for example, a channel that was on-air falls onto a closed hardware fader, the previous digital level before the scene change is maintained, and it is indicated by flashing on the OLED screen. If the operator touches the fader, the level is updated to the fader position.
- Audio will be muted if the input previously on-air is not assigned to one of the 18 channels in the new scene.
- Audio will be muted if the input previously on-air is assigned to a grouped channel in the new scene (groups always start off).
- The scene cannot be changed if there is a telephone line on-air or on hold. If attempted, a notification will appear stating that the communication must be ended first, and the scene change will be aborted.

**RENAME:** Allows changing the name of the selected scene. If the console is operating in the BASIC profile, this option is not available.

**COPY CURRENT:** Copies the currently active scene to the selected scene. A confirmation message appears for the action. If the action is not confirmed, it automatically cancels after 8 seconds. If the console is operating in the BASIC profile, this option is not available.

**SET TO DEFAULT:** Resets the active scene to a factory configuration (see 4.2 Default Routing of Inputs and Outputs). If the console is operating in the BASIC profile, this option is not available.

**SAVE:** Update the scene with the following data:

- Channel assignments to inputs.
- Channel assignments to groups.
- Signal assignments to outputs.
- Specific settings for inputs and outputs (phantom power, UNIDEX input options, MIX-MINUS modes, etc.).
- Input and output gains.
- Channel settings (labels, actions).
- Audio processor adjustments.
- Assignments to output buses.
- The scene does not save the state of the ON-AIR or CUE buttons.

### WARNING!

If any of these parameters are modified and the scene is not saved, the changes are lost when switching scenes.

If the console is operating in the BASIC profile, the SAVE option appears disabled.

Protected scenes (LOCK) can only be modified by the ADMIN user.

**5 CHANNEL MAP:** Launches a window that displays the configuration of all channels and allows simple reorganization through the following tools:

### SWAP 2 CH

It allows to swap the position of two channels:

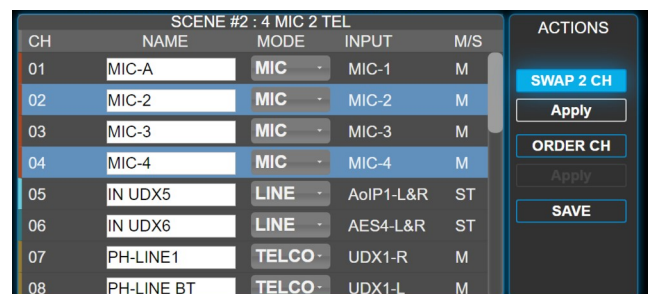


Image 40: Swap channels

1. Press SWAP 2 CH to enable the option.
2. Select the two channels you want to swap from the list.
3. Press APPLY. The channels will exchange their positions.
4. Press SAVE to make the changes permanent in the scene.

## ORDER CH

Allows redefining the position of one or more channels.

1. Touch ORDER CH. A dropdown menu appears to the right of the channel number, which allows assigning a new position for each one.



Image 41: Re-order the channels

2. Open the menu by touching the screen.
3. Choose a new position for that channel. Several channels can be changed at the same time.

WARNING!

If a channel position is already assigned to another input, the information in the destination channel is overwritten when applying the changes.

4. Press APPLY to confirm the changes. APPLY does not save the changes in the scene. If you exit the scene, the changes are discarded.
5. Press SAVE to permanently store the changes in the current scene.
6. To cancel the changes, deactivate ORDER CH by touching the option again or press the contextual BACK button.

## 5.10 VIRTUAL MIXER

UNIDEX UX18 handles 12 virtual channels. The 'VIRTUAL MIXER' screen allows the operator to control all the functions of the virtual channels



Pressing the encoder on the main screen accesses to the virtual channels. Screens for 6 channels are displayed

When one or more virtual channels are on-air (ON-AIR activated) the **VIRTUAL** item on the main screen continuously flashes in red to indicate this condition.

To facilitate touch operation, the channel controls are split into two screens:



LEVEL shows the faders and the ON-AIR button.



BUSES shows the assignment buttons PGM, REC, AUX, VBUS, and the CUE.

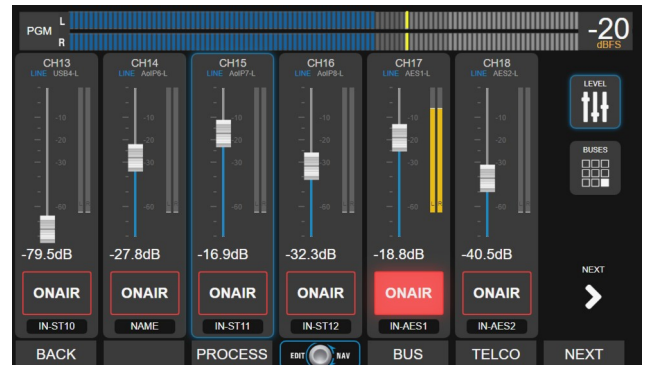


Image 42: Virtual channels mixer

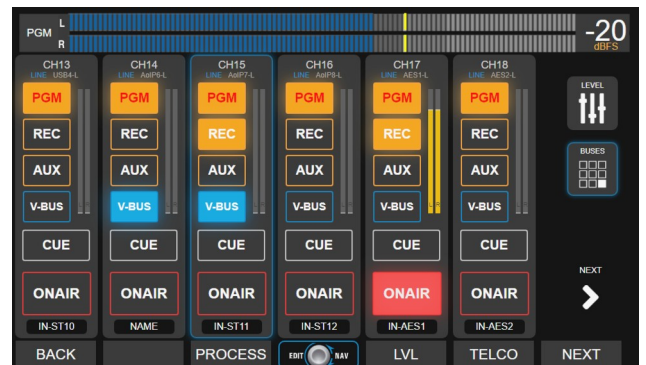


Image 43: Virtual channels assignments

### 5.10.1 Channels assigned to groups

The virtual channels can be assigned to a group channels, which are controlled with physical faders and buttons. There are four group channels that can have one or more virtual channels assigned to it.

Channels assigned to groups are displayed in the color corresponding to the group.

Assignments to buses are locked, as the channels inherit the group's assignment.

The power button of a grouped channel changes from ON-AIR to ON/OFF since the channel's on-air status depends on the activation of the group.

To manage grouped channels, access the GROUPS screen by pressing the shortcut button to the right of the screen.



# SECTION 6 WEB REMOTE INTERFACE

## 6.1 Remote access via IP

All functions and configurations can be remotely controlled by connecting UX18 to the local network. Access to the graphical control interface (HTML5) is achieved by entering the IP address into a web browser, from any terminal on the network.

### Step 1

Connect the ETHERNET Control port to a LAN router port using a standard Ethernet cable.

#### WARNING

Do NOT CONFUSE the Ethernet connector for CONTROL with the Ethernet port for STREAMING.

By default, UX18 operates with dynamic IP addressing (DHCP enabled). The router will assign it a free IP address. In this configuration, the assigned IP address may change when UX18 is restarted.

A static IP address can be assigned from the system settings screen (SETTINGS → SYS CONFIG). The as-

signed IP address is displayed on the main console screen.



Image 44: IP for remote control

### Step 2

From a computer connected to the same LAN (either via cable or WiFi), open a web browser (we recommend Google Chrome) and enter the Unidex IP address.

A login screen will appear, requesting a username and password. The operator can log in with either BASIC or ADMIN user. By default, the passwords match the username. The remote access user is independent of the user set in the UX18 hardware. Upon logging in, the following screen is presented:



Image 45: Remote WEB interface

### Step 3

- If the operator logs in with the BASIC user, only REMOTE CONTROL access is enabled.
- If the operator logs in with the ADMIN user, access to all operational and configuration functions is enabled.

#### Accessing via Internet

With the proper network configuration, it's possible to access the web interface from a remote location via the internet. This configuration requires advanced knowledge of network configuration.

## 6.2 WEB INTERFACE

On the left side, the web access screen presents the following options:

- HOME: Loads the current screen.
- STREAM CFG: Loads the settings options for the streaming. The options are the same as those on the onboard screen. See 3.7 STREAMING.
- REMOTE CONTROL: Loads the screen for remote operation and configuration (see below).
- SYSTEM: Displays options for configuring the local network for the REMOTE CONTROL Ethernet port.
- UPDATE: Loads the screen for automatically downloading firmware updates. See further in this section.
- REMOTE PLAYER: Allows listening to console audio over the network. See further in this section.
- LOG OUT: Logs out the user and returns to the login screen.

## 6.3 REMOTE CONTROL

Choosing the REMOTE CONTROL option from the menu provides access to the remote operation interface. The functions that can be controlled remotely are those available on the hardware and onboard screen, so the explanations provided earlier in this manual for each of them are valid. There are some operational differences described below.

The main screen of this section is similar to the "HOME" screen on the hardware. It displays level indicators, counters, "on-air" indication, and a menu at the bottom.

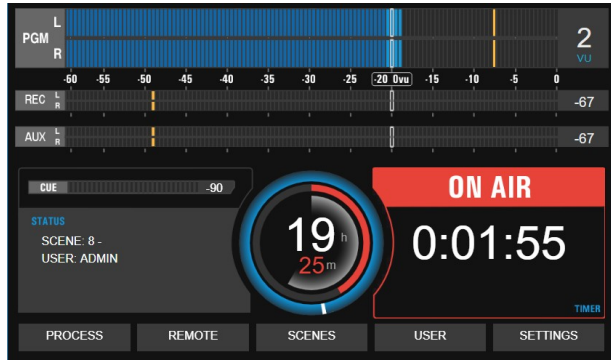


Imagen 46: REMOTE MIXER

**PROCESS:** Provides direct access to the processing screen, which can also be accessed from the "FX" option on the channels of the mix interface.

**REMOTE:** Accesses the mix screen, a graphical interface that emulates the controls of the control surface (see below).

**SCENES:** Allows switching between scenes. Note that switching the scene on the console is performed in the same way as if done from the onboard screen. There are no scenes specific to the web interface.

**SETTINGS:** Accesses the configuration options of UX18. The configuration options are organized in a vertical menu similar to the onboard settings screen.


### 6.3.1 REMOTE MIXER

The web interface can control all channels of the console. Its design is optimized for touchscreen operation. A common setup is to use the web mixer as an auxiliary controller to the onboard screen, running on a large touchscreen monitor.

At a more advanced level, the web mixer enables configuring an auxiliary studio to operate signals without the need for installing a second hardware console. For this scenario, the remote mixing interface features a monitoring panel and the ability to listen to audio sent by the console over the network.



Image 47: Remote WEB interface - Mixer

 The channels visible in the web mixer are set by clicking the "pencil" icon. Only the ADMIN user can modify this configuration. When you click the edit icon, a list of the 18 channels appears.

CHANNEL	NAME	MODE	INPUT	VIEW
01	MIC-1	MIC	MIC-1	<input checked="" type="checkbox"/>
02	MIC-2	MIC	MIC-2	<input checked="" type="checkbox"/>
03	MIC-3	MIC	MIC-3	<input type="checkbox"/>
04	MIC-4	MIC	MIC-4	<input type="checkbox"/>
05	MIC-5 UDX5	MIC	UDX5-L	<input type="checkbox"/>
06	MIC-6 UDX5	MIC	UDX5-R	<input type="checkbox"/>
07	PH-LINE1	TELCO	UDX1-L	<input checked="" type="checkbox"/>

Image 48: remote mixer channels

The checked channels will be visible, always in ascending order. Channels assigned to groups will remain hidden, becoming visible only when the corresponding group is expanded. These settings can be saved in the current scene.

### 6.3.1.1 Microphone and line channels

In each channel, the controls available on the UX18 surface are replicated. The operations performed on the web controller are reflected on the console surface.

**BUS BUTTONS:** Assign the channel to the bus.

**CUE:** Activates the CUE on that channel. The audio is heard through the CUE circuit on the console.

**ON AIR:** Turns the channel on/off.

**MAIN FADER:** Controls the channel level. When the level is modified from the web controller, the level changes on the console, and the value in dB is displayed on the OLED screen, which flashes to indicate that the level was modified remotely.

The physical faders do not change their position (they are not motorized). If the operator moves a fader on the console surface, the gain will match the new fader value, and the corresponding web fader position updates.

### 6.3.1.2 TELCO channels

The telecommunications channels include, in addition to the controls already seen, a RING/HOLD button that allows answering an incoming call and placing it on hold, as well as information about the line status.

The talk-back and dialing functions should be performed from the console or, in a remote studio setup, using an associated phone.

### 6.3.1.3 GROUPS

Channels assigned to groups do not appear in the general mixer. They are accessed by expanding the group, which displays the corresponding channels to the left.



Image 49: Channels 7 and 8 assigned to Group A

## 6.4 AUDIO PROCESSING

They are accessed from the PROCESS menu option on the REMOTE CONTROL screen; or from the FX icon of the channels in the WEB mixer.

- equalizer
- de-esser
- dynamics

**Remember that the processes are assigned to the inputs, so the available processes on a channel depend on the assigned input.**

*The usage is similar to that described for the onboard screen of the UX18 (see 5.9 Audio Processes).*



Image 50: Parametric Equalizer (web)

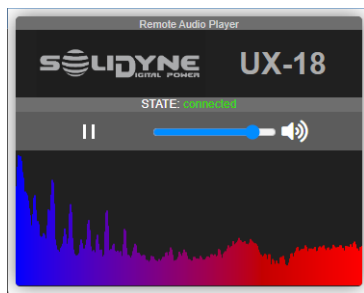
**1** The processors are inserted in each input. To view the processes of an input, select it from the INPUT drop-down menu. You can also choose a CHANNEL from the CHANNEL menu, and the processors of the input assigned to that channel will be displayed.

**2** The graphical interface of the parametric EQ presents some differences compared to the onboard screen. The ON/OFF buttons allow enabling or disabling each of the bands, while the checkboxes B1, B2, and B3 allow selecting a band to display its controls. This is done to minimize the number of controls and facilitate use on touch-screen displays.

**3** The different processes are toggled through these icons. Depending on the selected input, the available processors change. For more details, refer to section 5.9 - Audio Processes.

## 6.5 REMOTE PLAYER

The remote player allows listening to an audio signal sent by the console over the network.



The **signal source** is defined by accessing the main web access screen, under the STREAM CONFIG option → SERVICE CONFIG.

Signals such as PGM; REC; AUX; VBUS; STUDIO; CUE; or Talk-Back can be assigned (see 3.7.5 SERVICE CONFIG).

The **audio bitrate** can be defined by accessing the STREAM CONFIG option in the REMOTE PLAYER tab (see 3.7.4 REMOTE PLAYER).

## 6.6 FIRMWARE update

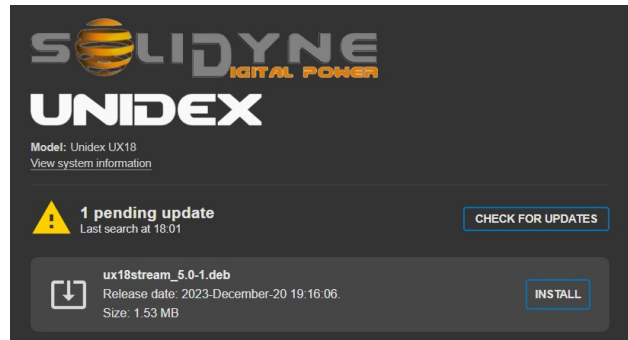
The console software is updated from the web remote control interface. Naturally, a computer with internet access is required.

Updates include:

- Mixing engine firmware (hardware)
- Onboard screen interface
- Remote web access interface
- Streaming services

To check for available updates and install them, follow these steps:

1. Ensure that the "Streaming Out" Ethernet port is connected to the network.
2. Using a computer on the network with internet access, open a web browser (preferably Google Chrome) and enter the "Stream IP" address (indicated on the console's main screen). The web access screen will open.
3. In the left menu, choose UPDATE.
4. The update screen will be loaded on the right side.



5. Click on "CHECK FOR UPDATES."
6. If there are no updates, the system will inform you that it is up to date.
7. If updates are available, a list with the details will appear.
8. The "INSTALL" option initiates the update process.

**The system NEVER INSTALLS UPDATES AUTOMATICALLY. Operator intervention is required to install an update.**

**Depending on the type of update, the unit may restart, MUTING the on-air signal for a few seconds.**



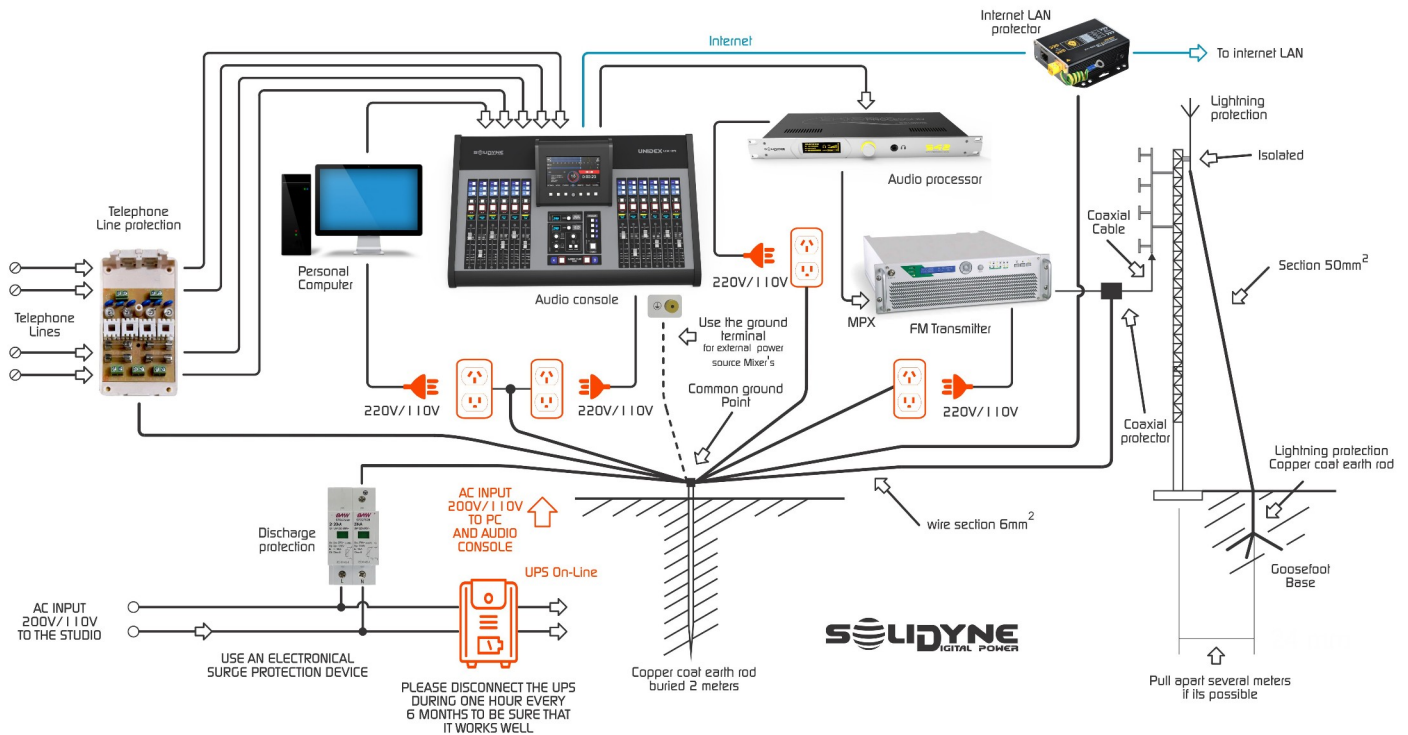
# SECCIÓN 7 DIAGRAMS and TECH NOTES

The following documentation is available at: <http://www.solidynepro.com/unidex-V>

## Diagrams

Recomendación general para la puesta a tierra de la emisora

[http://solidynepro.com/descargas/productos/Grounding\\_recommendations.jpg](http://solidynepro.com/descargas/productos/Grounding_recommendations.jpg)



## Quick Installation Guides

[Adapter UDX 2MIC](http://solidynepro.com/descargas/productos/UDX2MIC-Instalacion.pdf)

<http://solidynepro.com/descargas/productos/UDX2MIC-Instalacion.pdf>

[Adapter UDX 2TEL y 2TEL-P](http://solidynepro.com/descargas/productos/UDX2TEL-Instalacion.pdf)

<http://solidynepro.com/descargas/productos/UDX2TEL-Instalacion.pdf>

[Adapter UDX TEL+BT](http://solidynepro.com/descargas/productos/UDXTEL-BT-Instalacion.pdf)

<http://solidynepro.com/descargas/productos/UDXTEL-BT-Instalacion.pdf>

[Adapter UDX USB](http://www.solidynepro.com/descargas/productos/UDXUSB-Instalacion.pdf)

<http://www.solidynepro.com/descargas/productos/UDXUSB-Instalacion.pdf>

[DANTE/AES67 – Example of an implementation](http://www.solidynepro.com/descargas/productos/UNIDEX_UX18-A67-Dante_ESP.pdf)

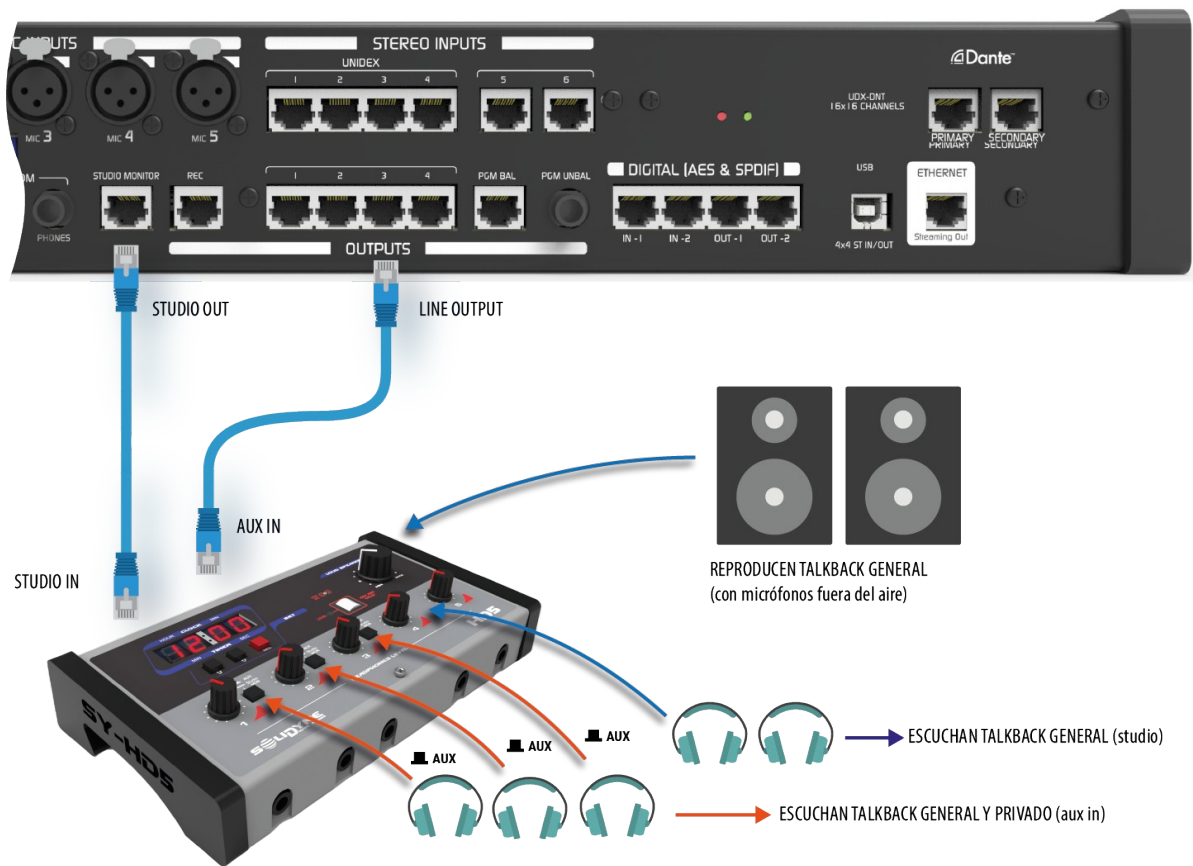
[http://www.solidynepro.com/descargas/productos/UNIDEX\\_UX18-A67-Dante\\_ESP.pdf](http://www.solidynepro.com/descargas/productos/UNIDEX_UX18-A67-Dante_ESP.pdf)

## Cables and wiring kits

Cables and adapters list (PDF)

[http://www.solidynepro.com/descargas/folletos\\_ingles/RJ45cablesUDX-EngCast.pdf](http://www.solidynepro.com/descargas/folletos_ingles/RJ45cablesUDX-EngCast.pdf)

## 7.3 Setting up a Solidyne StudioBox with double talk-back channel



The UX18 allows using the AUX input of the StudioBox to establish separate talk-back circuits. This allows the operator to speak to a group of headphones privately or to talk to all participants.

The previous diagram shows the connection and configuration of StudioBox HD5 with a dual talk-back channel. For the StudioBox HD3 case, the connection is the same.

### CONNECTION

- The STUDIO MONITOR input of the StudioBox is connected to the STUDIO MONITOR output on the UX18, using an STP RJ45 to RJ45 pin-to-pin cable (SOL P25 or P50 cables).
- The AUX-IN input of the StudioBox is connected to an assignable output of the console (OUT-1 to OUT-4), using an STP RJ45 to RJ45 pin-to-pin cable (SOL P25 or P50 cables).

### CONFIGURATION

- In StudioBox, assign one or more headphones to the AUX input.
- In the UX18, the output connected to AUX IN is configured as follows (see SETTINGS → OUTPUTS):

- **Source: STUDIO.** The signal delivered by the output follows the assignment made on the MONITOR screen.
- **Talk-Back Ch:** Select a channel or group to link it to the output. When Talk-Back is activated on that channel, Talk-Back will be sent, through this output, to the headphones assigned to AUX IN. For example: if you want to speak privately with the headphones of MIC-1, assign "Talk-Back Ch" to the MIC-1 channel.
- **Studio Talk-Back link:** Enabled.

### OPERATION

- With the above configuration, all headphones hear the signal assigned in the console for Studio Monitoring (at the MONITOR screen).
- The main Talk-Back button allows speaking to all headphones through the talk-back microphone.
- The CUE/SELECT button of the linked channel or group allows speaking ONLY to the headphones assigned to AUX. To do so, press and hold CUE/SELECT on that channel. The channel display will indicate TLBK.

# SECTION 8 TECHNICAL SPECIFICATIONS

## GENERALITIES

- 18 channels; 4 groups (sub-mixes) 10 hardware faders
- 12 Virtual faders operatives from the touch-screen
- Built-in WEB interface (HTML5) to remote control via LAN .
- Assignable inputs on the all chammes.
- 12 users scenes (input settings, channels assignment, audio processing)
- 7" touch.screen. 1024 x 600 pixels
- OLED in channels 1 to 6 (name; input level, gain; status)

## INPUTS (23)

- 5 MIC level inputs with 48V Phantom power.
- 3 stereo lines (2 bal, 1 unbal)
- 2 stereo AES3
- 4x4 stereo USB (four playing and four recording devices on Windows)
- 8 UNIDEX mono balanced (4 stereo) It supports external UDX adapters (mics, Telco devices, computers by USB)
- 1 Bluetooth to cell phones (1x Bluetooth - full duplex Hand free profile + A2DP music profile input)
- 16 DANTE AoIP (only for UX18/A67)

## OUTPUTS (23)

- 2 bus PGM estéreo (1 bal + 1 unbal)
- 1 bus REC (bal)
- 4 stereo assignable (unbal)
- 8 no balanced (support mix-minus)
- 2 stereo AES3
- 4 stereo USB (four recording devices on Windows)
- 2 independent audio streaming (RTP, Icecast/shoutcast)
- 2 stereo monitoring for the Control Room (loudspeakers and headphones)
- 2 stereo monitoring for the Studio (loudspeakers and headphones+talkback)
- 16 DANTE AoIP (only for UX18/A67)

## GPIO, INTERCOM AND AUXILIARY

- Tally Light output **12 VCC, 250 ma**
- 2 GPI (active high 3.3 ~ 15 v)
- 7 GPO (open collector; + 24V @ 250 mA)
- 1 unbalanced stereo for external tuner (assignable to channels)
- 1 talk-back input from studio (at the connector "Studio Box")
- Talk-back microphone
- Independent Intercom PTT in all TELCO channels with button CUE/SELECT.

## AUDIO SPECS

- 18 channels
- 4 stereo output buses (PGM, REC, AUX, VBUS)
- 1 bus CUE (PFL)
- 24bits SR@48 KHz
- DSP audio processing in all inputs. The audio processors are fixed on each input and their availability differs according to the input class. 3-bands parametric EQ. Noise gate. Dynamic compressor. Peak limiter. De-esser.
- Mix-minus sends and talkback/intercom (TELCO)
- **LATENCY:** 1,2 mS (analog line input to analog output) // 1,4 mS (MIC input to output using EQ + Dynamic + De-Esser)

## DIGITAL INPUTS AND OUTPUTS (AES3, AoIP)

- RD: 120 dBA
- Frequency Response: 20Hz-20KHz  $\pm$  0,1 dB
- THD: < 0,002 %
- AES3 IN samplerate: 44.1 / 48 kHz
- AES3 OUT samplerate: 48 kHz

## USB

- 1 USB 2.0
- 4 recording / 4 playing stereo devices
- Driver ASIO 2.2 (supports simultaneous play with ASIO and WDM)
- Supports 32-bit and 64-bit VST hosts.
- Supports multi-client (more than one ASIO app in parallel)
- Windows® 11/10/8/7 (32 & 64 bits)

## ANALOGICAL INPUTS

- Level: max: +26dBu, min: -24dBu , nominal: +4 dBu
- Dynamic range: 112 dBA
- Frequency Response: 20Hz-20Khz  $\pm$  0,25 dB
- Harmonic distortion: THD+N : 0,01 %

## MICROPHONE INPUTS

- Level: max: -24dBu, min: -75dBu , nominal: -50 dBu /150 ohm
- Attenuation (dip-switch): -10 dB MIC-1 to MIC-4.
- Noise: EIN @-50dBu input: -121 dBA
- Frequency Response: 20Hz-20Khz  $\pm$  0,5 dB
- Harmonic distortion: THD+N : 0,01 %
- Phantom power +48v. DIN 45596 compliant

## ANALOGICAL OUTPUTS

- Level: Nominal: +4 dBu Max: +24dBu
- Dynamic range: > 110 dBA
- Frequency Response: 20Hz - 20Khz  $\pm$  0,5 dB
- Harmonic distortion: THD: typ 0,005%@1Khz

## DANTE – AES67 (optional)

- DANTE 16x16 channels. Can work integrated to any DANTE network.
- Available output signals: PGM, REC, AUX, Control room headphones, Studio headphones and 3 stereo outputs assigned by the user.

## TOUCH SCREEN

- 7" Display 1024 x 600 pixels multi-touchscreen

## DIMENSIONS AND WEIGHT

- UX18 dimensions:
 

wide:	540 mm
deep:	435 mm
height:	82 mm
- Peso:
 

UX18=	6 Kg
UDX adapters:	0,33 Kg each unit
RJ45 wiring kit:	3 Kg
Con embalaje UX18 + Cables=	15 Kg