



**SOLIDYNE**  
DIGITAL POWER



COMPACT MIXER

**DX 816**  
**DX 822**

12/15 CHANNELS BROADCASTING MIXING CONSOLE + AOIP

OWNER'S MANUAL

[www.solidynepro.com](http://www.solidynepro.com)



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## About this manual

REVISION, June 2022

## What's in the box

Inside the packaging you receive the following items:

- 1 power supply.
- User's manual
- 1 guaranty agreement
- 1 tubular connector (2,5 mm) for Tally Light.
- 4 self-adhesive rubber pads.
- 1 Adapter patch cord to "Studio Monitor Out".
- **OPTIONAL:** Wiring kit for MNGX816.

When unpacking the unit; please check the contents to verify that the console has not received blows during the shipment.

## Introduction

Welcome to DX816/DX822 user Manual!





The Solidyne DX816 and DX822 are compact mixing consoles for broadcasting. Its design is solid and elegant; with a great flexibility. It includes USB digital I/O which brings digital link with the computer; avoiding the use of soundcards with the computer. In option, it can include up to two AoIP interfaces for PGM a REC.

Both models has professional "feather touch" *conductive ceramic faders of 100 mm*, with ETM-VCA control; with a life utility of 10 millions of operations.

This consoles are a great solution for small and medium radio stations; and for small recording studios for big broadcasters.

To take full advantage of the features of the console, we recommend to read carefully this user manual.

# WARNINGS

 <p><b>AC Voltage</b></p>	<p>This unit operates with an external switching power supply <b>110/240VCA</b>.</p> <p><b>OUTPUT 28 VDC</b></p>
	<p><b>MAKE SURE OF THE ELECTRICAL INSTALLATION HAVE A GOOD GROUNDING. USE THE GND COPPER BORN TO CONNECT THE DX822 TO GROUNDING</b></p>
	<p>The exclamation point within a triangle shown in this manual alert the user to the presence of important instructions on the operation and maintenance of the unit.</p>
	<p>The icon "pencil" appearing in this manual indicates the presence of an important note with information, suggestions and/or examples of use.</p>

# 1. Overview

## 1.1 Inputs

The broadcast mixing console **Solidyne DX816** has 5 microphone inputs, managed from 3 channel faders. The model DX822 has 10 microphone inputs, managed from six faders. The mic channels with dual inputs works with dual pre-amplifiers and audio compressors. The channels are labeled as follow (for the case of DX816):

Channel MIC 1 → Manages MIC IN 1

Channel MIC 2 → manages MIC IN 2 and MIC IN 2b

Channel MIC 3 → manages MIC IN 3 and MIC IN 3b

Channel MIC 4 → manages MIC IN 4

Channel MIC 5 → manages MIC IN 5 and MIC IN 5b

Channel MIC 6 → manages MIC IN 6 and MIC IN 6b

All main inputs are phantom powered (the second MIC IN 'N'b only works with dynamic microphones).

4 bands EQ and compressor/limiter of automatic action are available for the microphone channels. The action of dynamic compression can compensates differences of level when two MICs are used on a same channel.

4 line channels allow to connect up to 8 audio sources. The channels 4 and 5 have A/B switch to select between two sources (balanced and unbalanced inputs), and can receive incoming streaming from the optional AoIP modules (only models DX822/AoIP).

Two USB channels allows direct connection to the computer; plus aux inputs with 1/8" switched TRS.

The channel 'Telecom' manages 2 POTS (land lines) and one cell phone linked by Blue-tooth. Three calls can be on-the-air in conference. The use of "TELECOM" channel is easy and very intuitive. In addition, DX822 has a dedicated connection for external hybrids.

The models with the option AoIP can receive up to two incoming streaming via Ethernet, which are mixed to PGM and REC buses.

The available channels are:

- **DX816:** 5 microphone inputs over 3 faders. Channels 2, 3 manages dual mic inputs.
- **DX822:** 10 microphone inputs over 6 faders. Channels 2, 3, 5 and 7 manages dual mic inputs.

- **2 USB.**
- **2 balanced inputs.**
- **2 POTS**
- **1 cell phone (Bluetooth)**
- **I/O for external hybrid.** The signal is routed to Telecom channel.
- **In option, incoming streaming via AoIP.**

## 1.2 Outputs

Both models DX816/DX822 has two main outputs: PGM (program) and REC (recordings).

The channels USB 1 & USB 2 sends to the computer the signals PGM and REC. The computer will recognize it as USB recording devices.

In option, both models can have up to two AoIP modules, to send streaming of PGM and REC; and receive incoming streaming. This allows digital connection of DX816/DX822 with the transmitter plant, or with another Studios, using a LAN.

The outputs are:

- **PGM balanced.** The output **PROGRAM** is the main output, used for broadcast the audio to the audience (transmitter, streaming server, etc.).
- **REC balanced.** Used for recordings.
- **REC-USB.** The REC signal is also available on USB (when channel USB-1 is plugged). Usually used for recordings.
- **PGM-USB.** The PGM signal is also available on USB. It can be used to sent the signal to Internet streaming.
- **AoIP:** In models with the AoIP option, the signals PGM and REC can be sent through Ethernet, to STL link or interconnection between studios.

The mixer has send & return I/O for **external hybrid**. The audio from the external hybrid is routed trough the "Telecom" channel, and operates in the same way as the built-in hybrids (which continue being operative). Land lines connected to the mixer and the external hybrid can work in conference.

## 1.3 Monitoring

The section “Monitor” have the controls to route the signals to the loudspeakers and headphones of the Studio and the Control Room. The section also contains the controls for previous listening (CUE), talk-back and Master MIC-on. The talk-back microphone shares with the CUE circuit, to talk privately from the hybrid with calls.

Using the controls located in the section **Studio Monitor**, the operator selects the source of signal to be listened into the Studio (ON-AIR, PGM, REC or CUE). The mixer sends CUE signals to the headphones and loudspeakers monitors at the studio.

Into the Studio, a **headphone amplifier** (optional item Solidyne HD3 & HD5) is in charge of manage the signals of the loudspeakers monitors and headphones (see 2.5.1 - *Monitoring into the Studio* and 4.- *Monitoring dock Solidyne HD3 & HD5*).

The knobs at the “**Control Room**” section manages the levels for the headphones and loudspeakers. The switcher allows to select the audio source: ON-AIR (input for external tuner), PGM, REC or CUE.

**CUE:** The circuit CUE allows to listen the signal previous to the main fader (PFL). The signal CUE can be send to the Studio. At the Control Room, the operator can listen CUE on speakers or by headphones.

The output **Tally Light** gives 12 VDC when the Master MIC-on button is on. In addition, the console has an on-board tally light (see 2.5.3 - *Tally Light*).

## 1.4 GPIO

DX816/DX822 allows starting remote devices when certain channels are on-air (MIC-1; MIC-2; MIC-3; LINE 1 and LINE 2). It allows to command the on-air playout software; digital audio processors (like *Solidyne APC 542*); old CD/DAT/Minidisc players or others specific peripherals like the video switcher *Solidyne SmartCam*.

Models /AoIP uses Ethernet to send the signal “MICs on air” to the audio processor, in order to switch the audio processing preset to a voice profile.

## 1.5 ETM – VCA faders

The faders are “feather touch” with the technique of *control ETM-VCA* (Electrometric Voltage Controlled Amplifier). The faders DO NOT manage audio signals. The variation of the level is made by means of amplifiers of low noise and great stability. The main faders only handle control signals that modify the gain of the electronic amplifiers (please visit our web site for details). The main advantages of this technology are:

- *Eliminates the noisy signals due to dirty faders.*
- *Eliminates the maintenance and improves the life of faders.*
- *Gives a perfect stereo tracking (less than 0,1 dB error) between Left and Right channels.*
- *Allows using Conductive Ceramic faders of ten million guaranteed operations.*

# 2. Set up and connections

## 2.1 Overview

Installation of Solidyne audio console doesn't present particular problems. However, keep in mind the basic rules for all professional audio installations.



Do not connect the shielding of the audio wires with the general ground of the electrical installation. The console provides this union through a grounding terminal. See "2.3.4 - General diagram for grounding".

### 2.1.1 About the use of RJ45

The consoles come with their inputs and outputs with RJ45 connectors and are wired using multi-pair shielded cable CAT-5. With the advent of audio over IP (AoIP) many manufacturers started using RJ45 connectors and shielded multi-pair cable to replace different audio connectors, to standardize the entire in-

stallation with a single type of connector and a unique type of cable.

Additionally, the use of structured cable for connection between remote devices facilitates the installation anywhere in the world, due to availability of components and assembly tools used for data networks.

The multi-pair cable end that connects to the audio device (microphones, speakers, playback devices) still require standard audio connectors. For this reason, the RJ45 cable provides termination sections with RJ45 female at one end and a standard audio connector according the needs at the other end.





Figure 1: RJ-45-to-audio patch cords

All balanced inputs uses **RJ45 connectors and shielded twisted pair cables CAT5 (STP)**. The wiring kit Solidyne MNGX816/822 includes all cables, making easy the installation.

The pin out of RJ45 used for audio (balanced and unbalanced) are **full compatible with StudioHub** (<http://www.studiohub.com/>) with the exception of STUDIO MONITOR OUT.

## 2.2 Power source

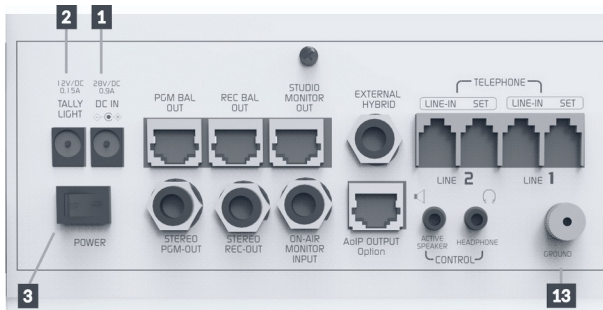


Figure 2: rear panel: power source

At the left side of the rear panel you find the connector for the power supply [1]. The switching power supply (provided with the unit) works with any AC network (90-240 VAC) and support great variations of voltage. Gives regulated 28 VCC



**IS RECOMMENDED TO USE ALWAYS THE ORIGINAL POWER SUPPLY**

The grounding made through a bronze tip located in the rear panel that provides the ground connection to the chassis. Use cable of 2mm to a good buried cooper bar.

## 2.3 Tally light

Tally light output ([2] in Fig.2) gives **12V/150 mA** when the Master MIC-on is turned on (supports two Solidyne LA-100 on-air sign). In addition, the console has a built-in tally light.



**THE OUTPUT "TALLY LIGHT" AUTOMATICALLY DISCONNECTS IF THE CHARGE EXCEED 150 mA.**

## 2.4 Audio inputs

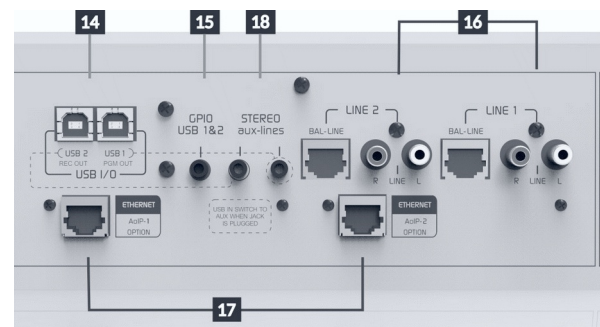


Figure 3: Inputs at rear panel

### 2.4.1 LINE INPUTS

The Solidyne DX816/DX822 has several inputs. Inputs '**BAL-LINE**' are *simetrycal balanced*. Inputs "**LINE**" are unbalanced. Inputs **DIG** are USB. The channels Line 1 and Line 2 gives balanced and unbalanced inputs [16] switched from the frontal panel. The USB channels have unbalanced aux inputs [18] using switched 1/8" TRS. **USB audio signal disconnects when a TRS is plugged.**

**Auxiliary inputs of the USB channels [18]:** The USB digital channels offer unbalanced stereo analog auxiliary inputs, using the 3.5mm (1/8") TRS plug. The auxiliary input is designed for the case in which the USB connection is not used, so the switching between the USB or auxiliary source is automatic: it occurs when a TRS plug is inserted in the input. While the TRS is connected, the USB audio remains muted.

The **balanced inputs** uses shielded RJ45 connectors. The wiring requires shielded twisted pair CAT5. In option, you can acquire the kit connection which include all cables needed for installation of the mixer console.

The following table shows the RJ45 pinout for balanced I/O:

RJ-45		RJ45 NOMENCLATURE	
PIN		PIN	CABLE COLOR
1	Orange / White	1	Orange / White
2	Orange	2	Orange
3	Green / White	3	Green / White
4	Blue	4	Blue
5	Blue / White	5	Blue / White
6	Green	6	Green
7	Brown / White	7	Brown / White
8	Broen	8	Broen

Table 1 - nomenclature RJ45

BALANCED INPUTS/OUTPUTS	
PIN	CABLE COLOR
1 Left channel (+)	Orange / White
2 Left channel (-)	Orange
3 Right channel(+)	Green / White
4 GND	Blue
5 Reserved	Blue / White
6 Right channel (-)	Green
7 -15 (optionally)	Brown / White
8 +15 (optionally)	Brown

Table 2 – balanced i/o RJ45

'LINE' inputs are unbalanced with "RCA" connectors. Equipment with outputs of -10 dBV @ 10 KOhms can be connected (home type or semi-pro).

The **GAIN control** is common for balanced line and unbalanced line (Line 1 and Line 2); for this reason, may be necessary to adjust the gain when switch the input.

## 2.4.2 MICROPHONE INPUTS

Microphone inputs are electronically balanced, using XLR connectors.

**DX-816:** has 3 MIC faders, but at the rear panel you'll find **5 XLR for MICs**. This is because inputs MIC-'N' and MIC-'Nb' share the pre-amp. This allows to manage simultaneously two inputs with one fader. Channel-2 manages MIC IN 2/2b and Channel-3, MIC IN 3/3b.

**DX-822:** Has 6 MIC faders. Like the 816, offer the option of use two mics over a same fader. Channel-2 manages MIC IN 2/2b; Channel-3, MIC IN 3/3b; Channel-5 MIC IN 5/5b and Channel-6 MIC IN 6/6b.

**If two microphones will be used on a same channel, they must be dynamic type and preferably of same brand and model.**

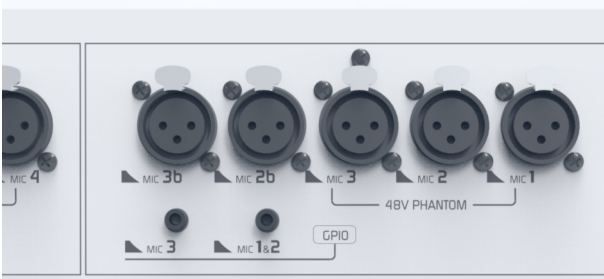


Figure 4: Microphone inputs

The button "48V" at the front panel enables the Phantom Power supply to the main mic inputs MIC-1, MIC-2, and MIC-3 (and mic 4, 5 and 6 for DX-822) . Phantom power is not available on MIC-2b, MIC-3b, MIC-5b and MIC-6b.

## 2.5 Audio outputs

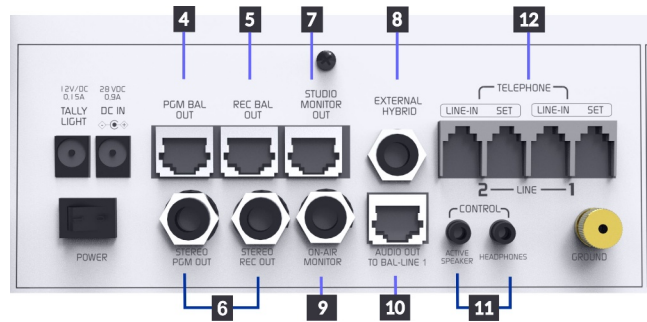


Figure 5: Rear panel: outputs, phone lines, AoIP

The main output is **PROGRAM (PGM)**. Is the mix signal that will be emitted. The PGM mix is available in several ways:

- A **RJ45 [4] connector** gives **balanced PGM**. Used to carry out the analog signal to the audio processor or to an STL. The audio processors Solidyne 362HD and 542APC comes with RJ45 balanced inputs, which simplifies the wiring. The pinout is the same used for the balanced inputs. Please refers to 2.3.1 - Line Inputs
- The jack **TRS [6]** gives **unbalanced stereo PGM**.
- **PGM also is available via USB** at the computer connected to Channel-6.
- In models with the option AoIP, PGM is available over Ethernet (PCM, MP3).

The **recording output (REC)** is available in multiple ways: balanced stereo on RJ45 [5]; unbalanced stereo on TRS [6]; via USB Channel-7 and over Ethernet in models with two AoIP modules.



**NEVER USE MONO PLUG'S (TS) TO CONNECT STEREO TRS OUTPUTS.**

The nominal output level is +4dBu @ 0VU. When the balanced outputs are connected as unbalanced, the level drops 6 dB, therefore 0VU = -2dBu.



If a device with an input level of -10 dBV is connected to the console output, it may happen that the output level is too high, causing saturation. In these cases, a fixed attenuator should be used to reduce the level (a resistive divider).

The "**Audio Out to Bal Line 1**" [10] output is only available on units equipped with the optional /AoIP. It is a balanced analog output that sends decoded audio from incoming streaming.



## 2.6 USB INTERFACE

The USB ports [14] are for direct connection to computers, using standard cables USB A/B. Ports 1.1 or 2.0 are supported.

Place the CPU next to the console in order to use cables of 2 to 3 meters of length. If needed, USB extension cables can be used; but it is recommended not to exceed 4 meters of length.



Figure 6: USB A/B



Always check the ground connection before connecting the USB.

### VERY IMPORTANT

- USB-2 CANNOT BE USED IF THE CHANNEL USB-1 IS NOT CONNECTED; since USB-2 is slave. In order to use only one USB channel, connect the channel "USB-1".
- WHEN TWO DIFFERENT COMPUTERS ARE CONNECTED; THE CHANNEL USB-2 CANNOT BE AVAILABLE WHEN THE COMPUTER CONNECTED TO USB-1 IS POWER OFF. Usually USB-1 is used for playout software; which is 24 Hs operative.

The channels must be connected to a computer running Windows®; starting by the USB 1.

**In case of only one channel was needed; connect always the USB 1.**

When connecting the USB channel to the computer, Windows® recognizes it and installs the correspondent drivers. Additional drivers are not required.

When plugs an USB channel, it appears in Windows® as "USB audio device" (play) and "USB recording device". In the console; the display "digital In/Out" will show "U4", indicating that stereo input and output of USB channel was recognized. When connecting the second USB channel the indicator will change to "U8"; and will appear in the computer a secondary playing and recording device (USB-2). Both USB channels can be connected to the same computer or to two different computers.

Check the playing and recording software settings to assign the new audio hardware. USB devices must be configured in the playout/on-air assistant software and in editing and others software that are in use. If the computer does not have installed another soundcards, default playing and recording device will be windows device USB-1.



### ABOUT USB DETECTION

We recommend do not change the USB cable to other USB ports, to avoid that Windows change the order of USB devices.

### 2.6.1 PGM and REC over USB

The USB channels provide two sends to the computer, which Windows © shows as "USB 1 recording device" (USB-1 = PGM) and "USB 2 recording devices" (USB-2 = REC).

This feature does not require any additional installation. USB recording devices appear when you connect the USB channels to the computer/s.

For details of use and configuration in Windows® see "3.6 - Program recording".

## 2.7 Monitoring

### 2.7.1 Monitoring at the Control Room

The outputs **CONTROL [11]** gives connection to loudspeakers and headphones, using 1/8" TRS Jack plug. The nominal level of the loudspeaker output is 0 dBu, to work with active monitors.

The operator has independent knobs to adjust the level of speakers and headphones; and independent switchers to select the audio source.



Control Room speakers are muted when the button Talk-back is pressed to avoid feedbacks with the talk-back mic.

### 2.7.2 Monitoring at control/studio unified

When the mixer console is used into a single room that merge the control room and the Studio, the audio monitors must be connected to the output **STUDIO MONITOR** (see next).

**The CONTROL (SPEAKERS) output is not muted** when the microphones are aired, so risk of feedback loops exists due to the mics and loudspeakers are into the same room.

### 2.7.3 Monitoring at the Studio

**STUDIO MONITOR OUT [7]** gives signal to the Studio Monitors using an **RJ45**. It allows to connect, using a standard RJ45 to RJ45 shielded patch (code SOL-P50) the monitoring docks **Solidyne Studiobox HD3** and **HD5** (optional).

**The speakers at the Studio are muted** when the mics are on air.

The headphones signal at the Studio is never muted. Studio headphones receives the signal assigned in console and the talk-back circuit.



**WARNING**  
Do not use audio RJ45 cables to connect the STUDIO MONITOR output. The pin-out used is not compatible with StudioHub wires.

For more information please see "4.- Monitoring docks Solidyne HD3 & HD5.

### 2.7.3.1 Monitoring setup without Solidyne StudioBox

In order to connect with any standard headphones mixer/amplifier and active loudspeakers; an adapter patch cord is supplied from factory.

The next diagram shows the available signals at STUDIO MONITOR OUT and their basic connections.

RJ-45		STUDIO MONITOR OUT	
#	SIGNAL	#	SIGNAL
1	Headphones LEFT (+4 dBu)	1	Headphones LEFT (+4 dBu)
2	Headphones RIGHT	2	Headphones RIGHT
3	GND	3	GND
4	ON-AIR (+14 V)	4	ON-AIR (+14 V)
5	Talk-back to Control Room	5	Talk-back to Control Room
6	VDC + 14 V	6	VDC + 14 V
7	Speakers LEFT (+4 dBu)	7	Speakers LEFT (+4 dBu)
8	Speakers RIGHT	8	Speakers RIGHT
	*Chassis = GND		

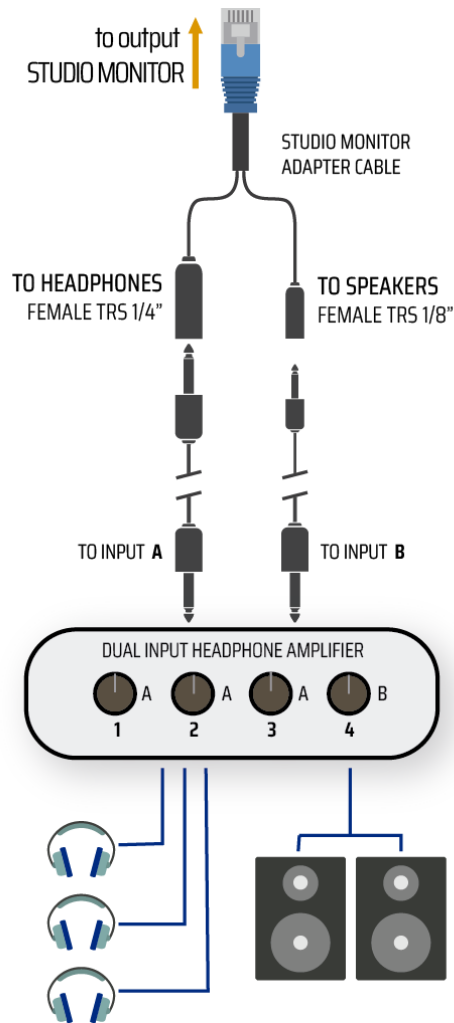


Figure 7: Connection of loudspeakers and headphones into the Studio using a standard headphone amplifier.

## 2.7 Connecting standard lines

The hybrid channel manages two land lines (POTS) and one cell phone linked by Bluetooth.

On the rear panel there are four RJ11: Two for the telephonic lines, and two for their associated telephones. The phone lines can be connected directly to public central or to a private central (PBX). The private central sometimes deteriorates the rejection factor of the hybrid, reason why we recommend to connect the console directly to the public central, whenever it is possible.

The **associate telephones** (used generally by the Production to answer the calls) are **connected to RJ11 labels as "SET"**.



**DO NOT PICK UP THE PHONE WHEN A CALL IS ON-AIR.** The telephone remains connected when the console take the line. If you pick up the handset, the phone will go live.

The telephone lines internally are protected against high voltage discharges, like non direct lighting, by metallic oxide varistors (SIOV). The internal protection protect until 2.000 V. Is strongly recommended to install external protection must be on POTS, grounded with a cooper javelin buried 2 meters or more. IN CASE OF LIGHTING, THE WARRANTY DOES NOT COVER DAMAGES. See "2.10 General ground diagram".



IS STRONGLY RECOMMENDED THE INSTALLATION OF ADDITIONAL LIGHTING/SURGE PROTECTOR USING GAS DISCHARGE TUBE (GDT) PROTECTION OVER EACH PHONE LINE.

THE WARRANTY DOES NOT COVER DAMAGES CAUSED BY LIGHTING OR OVER TENSION ISSUES.

### 2.7.1 Rejection factor (Null)

This adjustment is only for LAND lines. The rejection factor expresses the capacity of the hybrid to avoid that the transmitted signal returns distorted to the mixer. As greater this factor is, more "clean" will be the sound of the local speaker on the air. In order set the rejection, proceed as following:

1. Assign the TELECOM channel to REC.
2. Make a calling using the DX822's Hybrid. Someone into the Studio must dialog with the caller. Increase the level of fader Channel 8 until the voice of the caller reaches 0VU in REC vumeter.
3. Listening the voice of the local speaker in REC, turn carefully the preset NULL until find the point in which the sound quality is the same that the obtained with channel 8 off-air.
4. This adjustment change for each phone line, so if the line is changed, the adjustment must be repeated.

### 2.7.2 CONNECTING A CELL PHONE

Any Bluetooth cell phone can be linked to Solidyne DX822. While the phone can be up to 5 meters away from the console, is recommended leave it near to the console, **but not over the console**.

The procedure to link the phone to the console is similar to that used for other Bluetooth devices.

When the cell phone pair to the console, creates a link between the two devices, and the phone stores the ID of the console. This operation is performed only once (for each cell). Once the console and phone are paired, the console

automatically connects to the phone when the Bluetooth link is enabled on both devices.



DX822's BLUETOOTH IS POWERED BY USB CONNECTION. (CHANNELS 6 or 7). IF NONE USB CHANNEL IS CONNECTED TO A COMPUTER, CONNECT AN EXTERNAL USB POWER SOURCE TO ANY USB CHANNEL IN ORDER TO FEED THE BLUETOOTH.

Procedure:

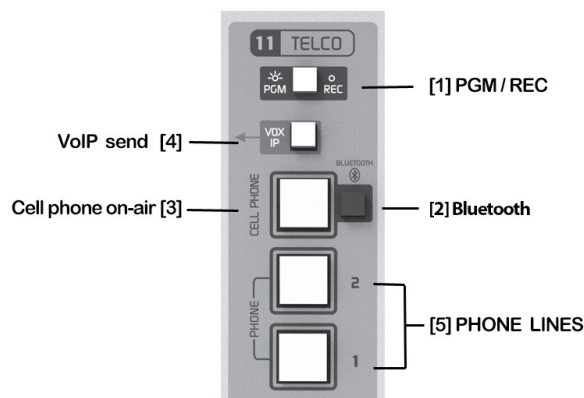


Figure 8: DX822 Telco channel

- **At the console:** Make sure that the LED of button CELL PHONE [3] is off. When Bluetooth is on, the button CELL PHONE flash in blue slowly (one flash every 2 seconds). In this case, turn off Bluetooth [2] pressing and holding until the flashing button turn off.
- **At the console,** enable the "discovery" mode. This mode only can be accessed if Bluetooth is turned off. Being the Bluetooth turned off, press and hold by 10 seconds the button BLUETOOTH, until the LED fast blinks in blue. It indicates that the console is in "Discovery" mode, to be recognized by the cell phone.
- **At the cell phone:** start a search for a Bluetooth device. This procedure can vary according to the brand and model of cell phone. Please refers to the correspondent users manual.
- When the searching ends, look for the DX816/822 Bluetooth device, which will appear as **Solidyne BTM641**. If the cell phone ask for a password, enter 0000 and confirm (see the user's manual of the cell phone).
- 4. The ID is now stored in the cell phone. The devices are "paired". You don't need to repeat this operation while using the same phone. The button now flash slowly in blue (one flash every 2 seconds), indicating "Bluetooth active".



On some phones you need to "connect" the new device found to stay active. In others, the new device is enabled after being detected.

### 2.6.2.1 Re-connect a paired phone

To reconnect to the console a cell phone, proceed as follow:

- Turn on the Bluetooth of the DX816/822 by pressing "BLUETOOTH" until the button CELL PHONE lights (5 secs aprox). The button will flash slowly in blue.
- Enable the Bluetooth at the cell phone. When the cell phone notice that the connection has been made you are ready to operate. When you make or receive a call, the audio is routed to the console.



In some phones the re-connection is not automatic. You need to "connect" the new device found to stay active.

### 2.6.2.2 Set up the level

**IMPORTANT: At the cell phone, Bluetooth volume must be adjusted at maximum level** in order to obtain a good reception and to sure the best signal to noise ratio.

**This adjusting must be done with a real calling using Bluetooth.**

The volume of the phone's speaker is not the same of Bluetooth device. If you change the phone's level without make a Bluetooth calling, you will only be changing the volume for the phone's speaker, not for the Bluetooth. To change the level of Bluetooth, make a call using the console (or a hands free headset) and set the Bluetooth level to the maximum, or one step below the maximum. This setting is stored at the cell phone's memory. If you use another cell phone with the console, you will need to set the Bluetooth level again.



### 2.6.2.3 Turn off Bluetooth at the console

To turn off Bluetooth, **press and hold "BLUETOOTH"** until the button CELL PHONE stop flashing.

## 2.7.3 External Hybrid

The DX816/822 series has send and return for external hybrid. "External Hybrid" [8] uses a female ¼" TRS (Jack) that send signal (PGM) through the "tip" and receives signal (return or input to console) through the "ring".

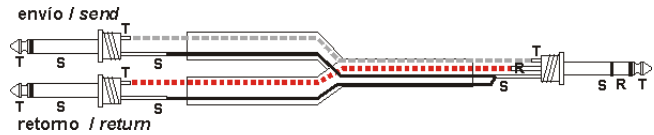


Figure 9: Cable for connect an external hybrid

**Ex-Hyb output (Tip);** Is the return from microphones from the console and PGM to the phone lines. This is a MIX-MINUS signal, and connects to the input "return to phone lines" at the external hybrid (audio input of the hybrid).

**Ex-Hyb input (Ring);** Is the input for the audio that in-comes from the external hybrid. This signal is mixed to PGM but not to MIX-MINUS signal. This input connects to the audio output of the external hybrid.

The audio from external hybrid is routed to TELCO (Channel 8) The main FADER of Channel 8 works in the same way that with the lines connected to the built-in hybrid.

## 2.8 GPIO

The 'GPIO' outputs (General Purpose Output) allows to command external devices when certain channels of the mixer are on-air. The channels that generates GPIO signals are: MIC-1, MIC-2; MIC-3 (and 4, 5 6 for DX822) USB-1 and USB-2.

These outputs are "open collector", and works like a switch: when the channel is off-air (fader closed or PGM unassigned), the output present an open circuit (high Z). When the channel is on-air, the output connects to GND, closing the circuit. GPIO can manage up to + 24V/0,1A.

GPIO use jack TRS 1/8" connectors, located at the rear panel.

GPIO MIC 1 & 2 (idem MIC 4 & 5 in DX822)	SLEEVE	GND
	RING	MIC-1
	TIP	MIC-2
GPIO MIC-3 (idem MIC-6)	SLEEVE	GND
	RING	MIC-3
	TIP	NOT CONNECTED
GPIO USB	SLEEVE	GND
	RING	USB-1
	TIP	USB-2

Table 3 - GPIO

## 2.9 Models /AoIP (optional)

The models DX816/AoIP and DX822/AoIP can include up to two Ethernet interfaces. See Figure 3.

/AoIP allows to stream the PGM signal (models with only one AoIP) or PGM and REC (models with double AoIP). The AoIP module allows:

- Link with the transmitter plant (STL). At the transmitter plant a dedicated hardware receive the stream (Solidyne ADA102 or processors Solidyne 562dsp/IP or series 542 APC).
- Send the PGM signal to another computer of the network. This computer can decode the streaming using any audio player software with support for RTP streaming (i.e. VLC Player).
- Link with another Solidyne AoIP console ( D612; DX816/822 /AoIP; Series 2600 and UNIDEX UX24).

### 2.9.1 AoIP power source

The IP module receives power from the USB connection, so it is only available when the console is connected to a computer via USB.

If the console is not connected to a computer via USB, a standard USB power source (5V) can be connected to DX816/822.

### 2.9.2 Studio to Transmitter Link

In studies, the /AoIP console works as ENCODER, generating an audio streaming to establish a unidirectional link (half-duplex) between the Studio and the Transmitter Plant. At the transmitter plant the streaming is decoded by a dedicated hardware (i.e. Solidyne ADA102 or audio processor with streaming AoIP).

### 2.9.3 Set up the AoIP module

Setup options are available in an embed web page. By default, the unit works with "Dynamic IP". Connecting the console to a LAN, the unit gets an IP address via DHCP (the router will assign an IP). The procedure is as follows:

#### Step 1

Connect the unit to the network using a standard cable. The network must have a router. DX816/822 with AoIP can also connect directly to a modem-router, since usually it will assign an IP via DHCP.

Once the IP is obtained, a green LED at the rear panel (RJ45) stays blinking.

#### Step 2

To know the IP address, install and run the tool "Solidyne Multi-discovery", available at:

[solidynepro.com/DW/IP.exe](http://solidynepro.com/DW/IP.exe)

The file IP.exe is a software setup that installs the "Solidyne Multi-Discovery". Look for leame-readme.txt and follow the indications according the case.

#### Step 3

Multi-Discovery will show the device in a list box. Double clic on it to open the AoIP (streaming) web interface. Or you can open a web browser and enter the IP address showed by Multi-Discovery.

## 2.9.4 Streaming settings

### 2.9.4.1 Set the destiny IP

The screen STATUS indicates the IP port configuration. By default the module is set as Studio Encoder (option "Location" → 'Studio Encoder').

#### Step 4

To check the settings, access to "Configuration" and choose "Basic settings".

BASIC SETTINGS

OUTGOING STREAM

Stream Method	URL	Port
Push(RTP)	192.168.0.30	4050

- "Stream method" must be "Push (RTP)".
- In the URL field defines the DNS name or IP address, and the destination port to which the console streaming broadcast.
- Press "Apply" to confirm the settings.

The configurations listed below require knowledge in network administration.

The destination IP address is the external address of the network in the Transmitter Plant, where it is connected the DECODER (static IP assigned by your ISP). When the data packets reach the router/firewall on the other end, must be re-directed to the DECODER IP (eg 192.168.0.30).

Since the mixer transmitt to a specific IP address and port, at the transmitter plant all packets that arrives at this specific address and port must be forwarded to the DECODER, or a particular computer, which will decode the stream as audio signal. To identify which packets must be sent use *port forwarding*.



## 2.9.4.2 Audio options

### Step 5

Go to *Config* → *Audio*

#### Input source

Default is “Stereo line”. Don’t change it.

#### Format

Audio settings for the streaming. Default values are:

Format: **PCM 16 bits stereo**    Sample rate: **48 KHz**

#### Studio Transmitter Link

Studio Encoder / Decoder AUDIO

Format: PCM 16bit stereo MSB

Sample Rate: 48 kHz

Input Source: Line Stereo

Input Gain: 0 dB

Output Audio Level: 0 dB

Output Audio Level Offset: Auto

Fading Effect:  Enabled  Disabled

Apply Cancel

PCM16 @ 48KHz generates a streaming of 1.6 Mbit/s.

## 2.9.5 Using a digital microwave link

An /AoIP console connected at the studio to a broadband Internet, can cover any distance from the studios to the transmitter plant.

For short distances, a good solution is to use a point-to-point RF data link to carry out uncompressed audio (PCM 16 bit/48 KHz). For this, install a microwave link for 5.8 GHz (or 2.4 GHz in some countries) under the standard 802.11.x. This band is available in all countries and does not require special authorization. It can cover up to 27 miles if there are no obstacles between the extremes. Encoded audio can be transported for special applications that requires a bidirectional link.

For more details, please contact us at [info@solidynepro.com](mailto:info@solidynepro.com).

## 2.9.6 Receive an incoming streaming

The firmware can be settled to decode an incoming streaming; for example, a streaming generated from other Solidyne device (series 2600/IP; DX816; DX822; Unidex UX24 and ADA102 coder). The encoder unit and the console must have the same firmware (STL).

From factory, DX816/822 is configured to work as coder, to generate the streaming, To receive a incoming streaming, the mode of work must be changed to DECODER.

To set this mode, proceed:

1. Using any WEB browser, access to the IP module (please refers to 2.8.1.1 Access to IP module ).
2. At the top menu, click the option “*Location*”.
3. By default, the mixer works as CODER, generating a streaming (mode “Studio Encoder”). Change the mode to ‘Transmitter decoder’. When the unit are configured as decoder, default values for reception are loaded. Usually the user don’t needs to change nothing, with the exception of PORTS and IP ADDRESS.
4. Enter to the option “*Settings* → *Streaming*”.

#### OUTGOING STREAM

Output Trigger Level: 1000

Output Inactivity Timeout: 1000 msec

Keep-alive Period: 1000 msec

INCOMING STREAM

Stream Method	URL	Port
Push(RTP)	0.0.0.0	3030

RTP delay: 200

5. In the field “*Port*” enter the same port number used in the CODER side. Hit “Apply” to confirm.
6. In “URL” leave the default value 0.0.0.0. Press “Apply” to confirm.

*At the decoder, pay attention to the buffer size, which is the memory that holds the incoming streaming audio to avoid drops. These interruptions occur when the bandwidth is insufficient for the transmitted audio quality and can be solved by increasing the buffer size. But note that larger buffer size will cause greater delay time. The appropriate value depends on the bandwidth of the network and streaming codec. Buffer size is expressed in milliseconds. For uncompressed PCM streaming times of 40/80 mS works find. For coded formats larger times are needed due to the times involved in the decoding process.*

No other value must be changed. The console automatically recognize the incoming streaming. The audio quality is defined in the CODER side.

### 2.9.6.1 Audio from incoming streaming

The console receives the streaming through the AoIP module. The incoming streaming is decoded and sent to a balanced input **using an external patch cord supplied from factory** (standard Ethernet RJ45 to RJ45).

This patch cord connects the analog output “Audio Out to Bal-Line” (see item [10], Figure.6 at “2.4 Audio outputs”) with a balanced input.

For the case of models with two AoIP modules, the output “Audio Out to Bal-Line” connects to both balanced inputs, using a special patch cord RJ45 “Y” supplied by Solidyne.

## 2.9.7 Receiving with a computer

To receive the streaming from the DX816/822 /AoIP mixer a computer, use any playback software that support streams with RTP protocol (i.e. VLC Player <http://www.videolan.org/vlc/>).

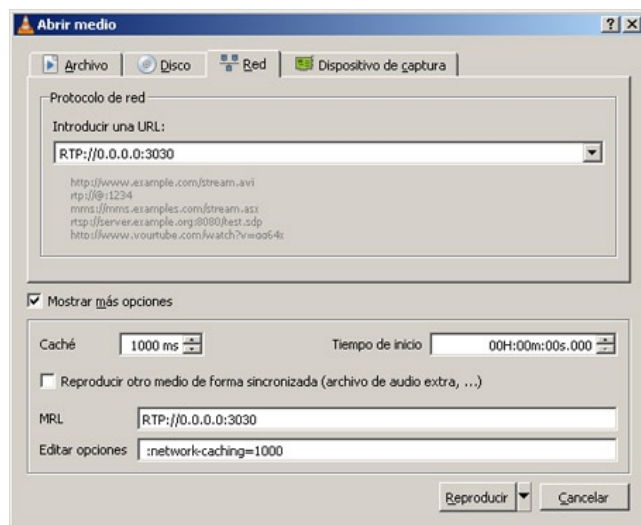
To start to play in VLC go to Media -> Open network

In URL enter: [protocol]:/[sourceIP]:[port]

For example: RTP://0.0.0.0:3030

In where RTP is the protocol used by the Solidyne DX822/ AoIP; 0.0.0.0 means from any incoming IP and '3030' is the port used. Remember to set the firewall to allow incoming streaming trough port 3030 (or the port that you use).

At the console, set the transmission to **PCM16@44.1KHz**; or any MP3 mode (see 2.7.1.3 - Audio settings).



NOTE: Some versions of VLC Player also supports the string RTP://@:3030

## 2.9.8 Firmware update

### Connection with shoutcast / icecast servers

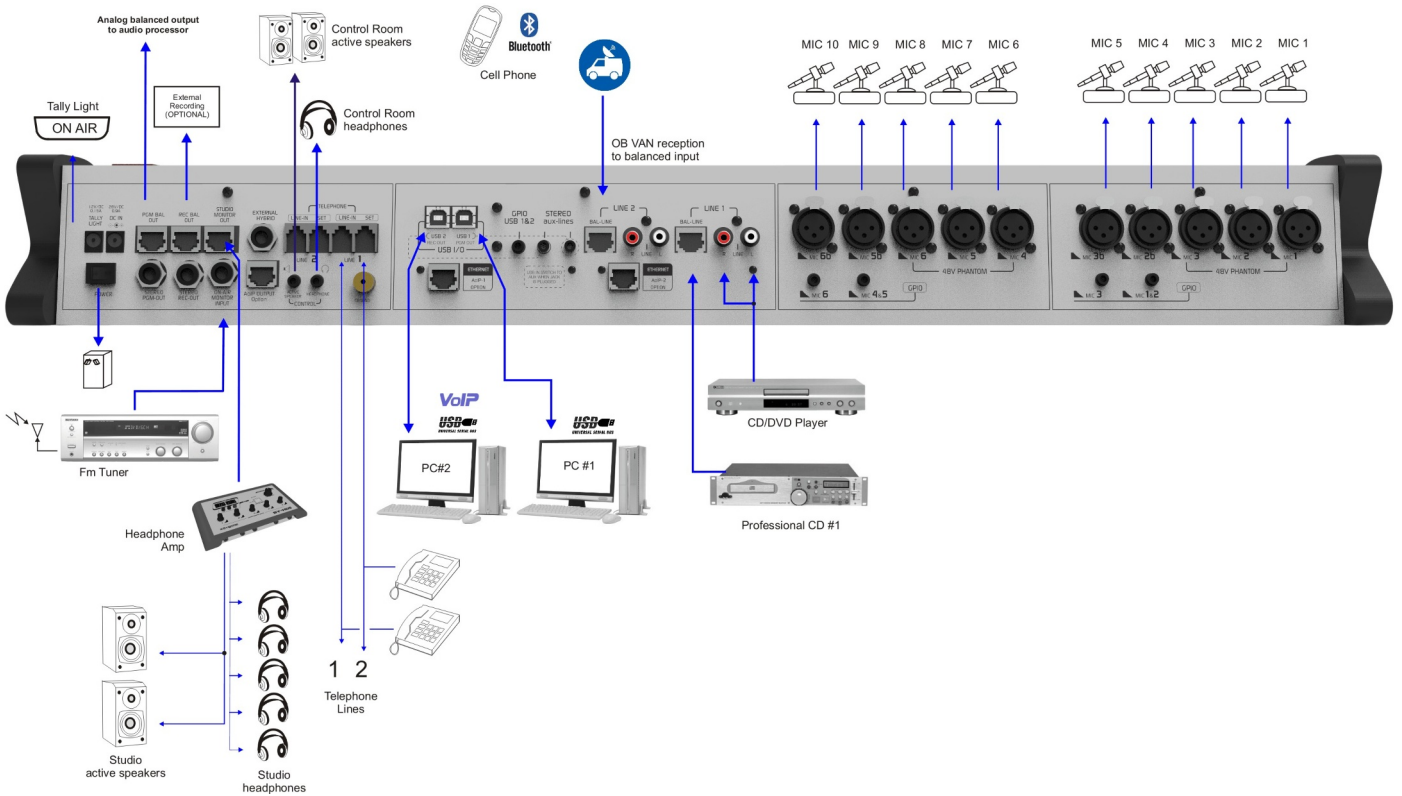
All modules / AoIP come from the factory with STL firmware, for studio link to transmitter plant.

The firmware can be updated to:

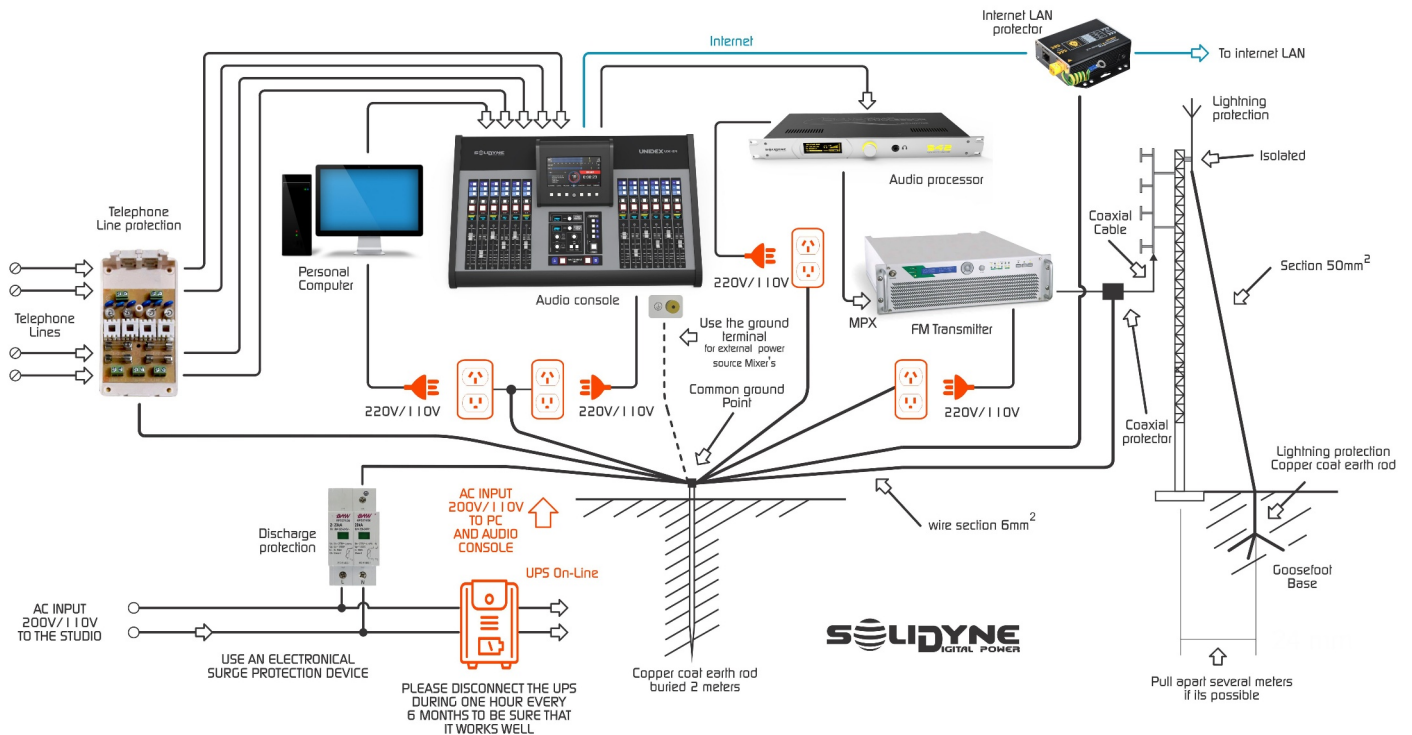
- Upgrade to a new STL version
- Change the STL device to the Upstreamer model, for Internet connection with shoutcast / icecast servers.

The procedure for changing firmware and the Upstreamer mode configuration are detailed in the following document available on our website: Technical Info → Tutorials and technical notes or click on the following link [“Firmware updates and software connections”](#)

## 2.10 Interconnection diagram



## 2.11 Grounding connection diagram



## 3. Operation

### 3.1 Overview

The console desktop presents several areas:

- *Input channels.*
- *A telecommunication channel.*
- *Monitoring and talk-back.*
- *Previous listening (CUE)*
- *Level indicators.*

The console generates two output mixes:

- **PGM output** sends audio to the transmitter plant, or is used to streaming.
- **REC output** for recordings with independence of PGM signal.

The buttons PGM and REC assign each channel to the outputs (PROGRAM and RECORDINGS).

The operator can listen an audio source before put it on-air, by pressing the CUE button (Pre Fader Level). CUE hears on headphones and optionally, on the main speakers of the Control Room.

To put on-the-air a LINE channel, assign it to bus PGM and raise the fader. To put on-the-air a MIC channel, assign it to bus PGM; raise the fader and press the button MASTER MIC-on.

Microphone channels have a switch to turn on the 48V phantom power source. The phantom power enables for the inputs MIC-1, MIC-2 and MIC-3 (not available for inputs MIC-2b and MIC-3b). In models DX822 a second switch turn on/off the phantom power for the inputs MIC-4, MIC-5 and MIC-6 (not available for MIC-5b and MIC-6b)

Some microphone channels can manage two mic inputs at the same time (MICS 2 & 3 at DX816; and MICS 2, 3, 5, & 6 at DX822). See 2.3.3 *Microphone Inputs*.

If two microphones will be used connected on a same channel, these microphones must be dynamic type and of same brand and model.

The equalization is global for all microphones.

Dynamic compression applies only over the MIC channels.

The **USB** channels connects directly to the computer, managing two stereo signals. In addition; **USB send to the computer** the mixes PGM (channel USB-1) and REC (channel USB-2).

DX816/822 /AoIP can have up to two Ethernet modules (AoIP) for send/receive audio over IP (PGM and REC). Each AoIP module can receive a

streaming which is routed to an input channel of the mixer.

The turret contains electronic LED's VU-meters that show the recording (REC) and program (PGM) level (average level). A Dynamic Compression indicator shows the gain reduction applied to the mic signal.

#### 3.1.1 Quick checking

Next are described the basic procedures to verify the console wiring. In order to check that all is working well, please follows the next steps:

1. Choose a signal source, like a microphone, a CD player, etc. According to the used source, select in the channel the correspondent input (MIC-LINE or BAL-LIN). This can be the first cause of error (there is no signal because the mistaken input is assigned).
2. Press CUE button in the channel in which the signal is.
3. Open the CUE fader in the MONITOR SECTION. The audio will have listened in the built-in loudspeaker.

On the air:

1. Send the channel to PGM, so that the signal is sent to the main output.
2. Open the main fader until reach 0 VU in program VU-meter. The signal will be on the air.

#### NOTICE

The gain knob must adjust so that the normal work position of the main fader is -15 dB (gray zone). In some countries (like England) the user prefers to adjust the normal level with the fader at maximum.

### 3.2 Hybrid & TELCO

The Channel TELECOM manages 2 land lines (POTS) and 1 Bluetooth cell phone. The three lines can talk to each other in conference mode. It's also possible put in conference a VoIP communication (like Skype). VoIP callings are made using a computer connected to Channel USB-2. By pressing the button VOX-IP [4], TELECOM channel sends to the computer the return signal from the callings, through bus REC.

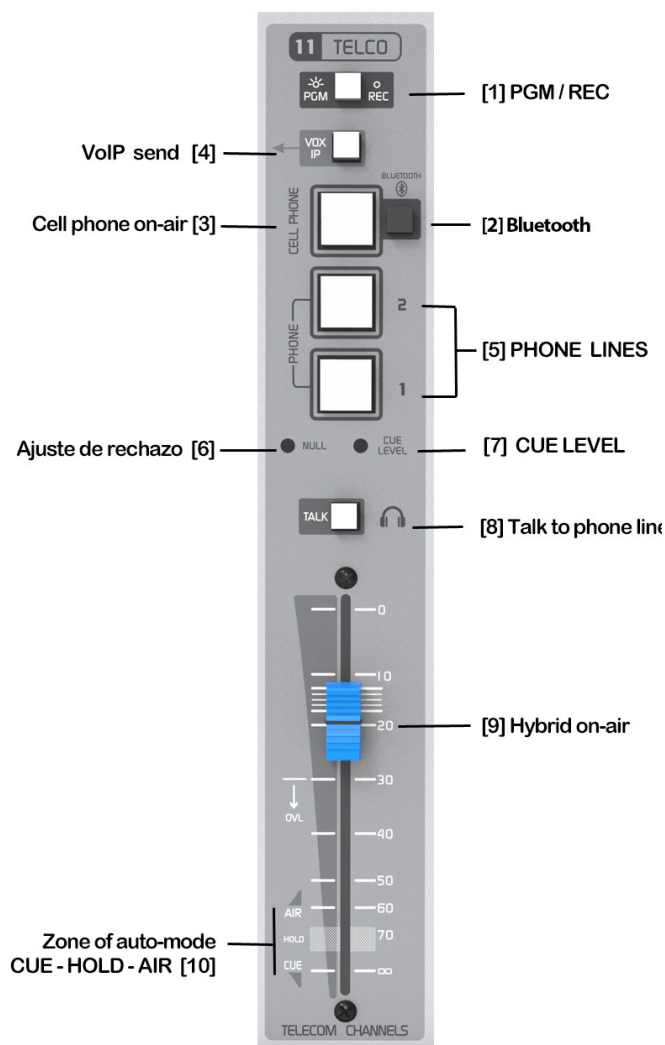


Figure 10: Telecom channel

#### VERY IMPORTANT

- USB2 CANNOT BE USED IF THE CHANNEL USB1 IS NOT CONNECTED; since USB2 is slave. In order to use only one USB channel, connect the channel "USB1".
- WHEN TWO DIFFERENT COMPUTERS ARE CONNECTED; THE CHANNEL USB2 CANNOT BE AVAILABLE WHEN THE COMPUTER CONNECTED TO USB1 IS POWER OFF. Usually USB 1 is used with the air PC; which is 24 Hs operative.

### 3.2.1 Using the TELECOM channel

For this explanation, we define two modes of work:

- A DJ or Talent that answer itself the callings of the audience (small stations; also used by many radio stations outside rush hour).
- A producer/telephonist answer the calls.

#### 3.2.1.1 Case A: A DJ or Talent answer the calling of the audience

The mixer has a telecommunication channel that manages up to 3 phone lines (one is a cell phone linked by Bluetooth).

- 1) The DJ/Talent can manage the incoming calls unassisted. When a calling incomes, the button of that line blink with the 'ring' cadence.
- 2) To answer a call, press the correspondent blinking button. The button will light in red. For the case of cell phone, press the button CELL PHONE [3] to answer.
- 3) If there are more than one line in red (waiting in HOLD), when the fader is raised to on-air zone all lines go on-air in conference.
- 4) When a call is answered, the TELECOM fader [9] must be closed, in the position CUE [10]. In this condition, the caller remains hearing the program mix (the audio that is on-air), and the audio from the phone line sends to the CUE bus.
- 5) To talk to the line(s), press and hold the button TALK [8]. While TALK is pressed the speakers of the Control Room remains muted, staying the monitoring on headphones.
- 6) The button TALK [8] is retained. With the main fader in CUE position, the operator can talk hands free. The caller listen on headphones (since the loudspeakers remains muted to avoid feedback loops).
- 7) If the main fader is into the zone labeled as "HOLD", the calling is HOLD waiting to be on-air. The caller hears the PGM signal. In this mode, the phone line is not send to CUE.
- 8) To put a calling on-air (or more callings in a conference) raise the main fader [9] until achieve the apropiate level on VU meters. All lines that have the button pressed (red) will be on-the-air.
- 9) To end a call first close the main fader and then release the button of the line. For the case of CELL PHONE [3] the call is transferred to the cell phone, and can be re-taken by pressing CELL PHONE once. To hang out the cell phone, short touch the button BLUETOOTH [2].

#### 3.2.1.2 Case B: The producers team or telephonist answer the calls

- 1) In the Production Room there will be two phones "Phone-1" and "Phone-2", each one for a phone line in the console; and the mobile phone.



- 2) When a calling is answered or made, Production inform to the operator in order to he take the line, by pressing the correspondent button on the console. When the operator takes the line, the Production can hung the phone.
- 3) All calling answered or generated from the cell phone can be transferred to the console using the cell phone. For make this, set the Blue-tooth mode to "Headset" and advise to the operator, who must press CELL PHONE [3] to put the call in hold (the caller hear the PGM signal, but is not on-air).
 

**Note:** If you needs to work with several call in hold mode while another call is on-air (and the caller waits hearing the on-air signal) you needs to add an external hybrid. For this purpose the Solidyne DX mixers have a special input/output (see 2.2.8 – External Hybrid).
- 4) Being one or more phone line buttons pressed, (in red) press TALK to dialog with the callers. When TALK is active, the speakers are muted at the Control Room.
- 5) If the operator leave pressed the button TALK, can dialog in "hands free" mode with the caller, and listen he on headphones (since loudspeakers are muted to avoid feedback). The level of the TALK MIC adjust from a preset located next to the TALK microphone.
- 6) To put on-air a phone line, raise the main fader TELECOM [9].
- 7) To hang out a line, first close the main fader and then release the button of the line. For the case of CELL PHONE [3] the call is transferred to the cell phone, and can be re-taken by pressing CELL PHONE once. To hang out the cell phone, short touch the button BLUETOOTH [2].
- 8) To return a calling to the Production (i.e. to take note about a listener) the Production must pick up the phone and then the operator will release the correspondent button at the console. If the line is the cell phone, disable the mode "hands free" at the cell phone to resume the communication on the cell phone.



The red LED in phone line buttons only lights with active phone lines. If the line is not connected, the LINE button not lights.

#### TAKE IN MIND

Please be careful with the on-air level of the callings. Due to the analogical hybrids are not perfect, a brief amount of audio sent to the line is not cancelled, returns through hybrid and mixes to the direct signal (the vocie from studio). Working with normal levels this effect is not noticeable; but using high hybrid output level, the voices of the studio can be distorted (by coloration).

A phone calling not always must achieve 0 VU to equal the loudness of the voices from the studio. Have in mind that

the LF, that carries greater energy, is not present in the telephone line, but there is in the voices at the studio. The lows are those that produce greater deviation in VU METERS. Nevertheless, for the ear the loudness is defined by the mid rage frequencies. So that if you note that VU Meter "peaks lower" with the telephone line that with the voices from mics on the studio, this not necessarily means that "it's listened lower". On the other hand, the audio processor of the radio will be in charge to equal both signals, so you do not leave the VU deceives to you when mix voices of the floor with telephone calls.



#### VERY IMPORTANT NOTE

Conference between land lines only are possible if both lines are of the same public central. The lines can be connected to the same private telephonic central (be of 24 or 48 volts). The conference mode IS NOT POSSIBLE BETWEEN DIFFERENT CENTRALS neither different IP converters.

If the voice of the Studio Talent is colored **when the hybrid is on-air**, (sounds too much mid-lows or flanges); the rejection factor of the hybrid needs to be adjusted. See "3.2.5 Rejection factor".

If the calling at CUE listen distorted, or with excessive ambience, the level of preset **CUE LEVEL [7]** is excessive. Attenuate this level until reach an optimal listening of the MICs at the Studio.

### 3.2.2 Return to phone line

Most of the existing telephone hybrids on the market today, were designed over 30 years ago for analogue telephone exchanges (PBX) Solidyne hybrids, however, have been recently designed for private or public telephone exchanges today, which are fully digital. The new technology Hybrids are recognized because they have no control of air return level to phone line. This is because inside the hybrid Solidyne uses an audio processor for return signal that includes AGC, peak limiter & audio signal filtering. Therefore the return is automatically adjusted during the transmission and its level is the maximum allowed by the modern digital telephone exchanges.

If you want to check the return level to phone line, you must use a oscilloscope to be placed in parallel with the telephone line and must verify that the signal is 2 volts peak to peak.

Please note that above this level the return can produce problems that will cause intermodulation distortion in the audio signal that goes to air. So in Solidyne hybrids we use a processed return channel, to avoid distortion at the on-air signal. There are hybrids manufacturers that maintain the return control level as they did in the past. This allow operators to adjust "by hunch" this critic level. This makes the voices of the reporters and interviewed people distorted or with coloration.

In Solidyne obviously, we keep a high grade of excellence in the audio quality of the hybrid on-air sound. And that quality do not depend on the operator set-

tings. Note that the Solidyne Hybrid on-air audio quality of the local journalists is ever perfect and without any coloration.

To achieve this level of quality we use a narrow-band return filter. Then the return signal is limited to the band 400 – 2.200 Hz in order not to distort the signal to the air. This narrow band intelligibility remains high (due to processing) but occasionally may seem to the remote people that it "has little volume" because his band is narrow. This should not worry because it is a subjective sensation that does not affect the intelligibility of speech.

### 3.2.3 'PGM/REC'

**So that the communication can be on air, the switch PGM-REC must be at the position PGM.** Please see "3.6.1 - Recording of telephone lines".

**IF YOU DO NOT HAVE AUDIO ON-AIR FROM THE HYBRID; CHECK THE POSITION OF THE SWITCH "PGM-REC". THIS MUST BE "PGM"**

### 3.2.4 Using with cell phone

If the cell phone will be used with the mixer for first time, the devices must be paired. To know this procedure refers to "**2.6.2 – Connecting a cell phone**".

Once paired, the procedure used daily is very simple:

- Make sure that the fader TELECOM [9] be closed (CUE zone).
- Turns on Bluetooth at the console. by pressing and holding the button BLUETOOTH [2] (2 seconds approximately) and release it just when the button CELL PHONE [3] lights. The button CELL PHONE remains blinking in blue slowly to indicate that the Bluetooth link is enabled.
- Turns on Bluetooth at the cell phone. If the cell phone was paired with the DX822, the link will be established and in few seconds the phone is linked with the mixer. In this condition, the incoming calls are transferred directly to the DX822.

#### 3.2.4.1 INCOMING CALLS

Next is detailed the use of a cell phone in permanent link to the console. In this condition, the procedure to put a calling on-the-air is similar to the seen for land lines:

- When a call incomes, the 'ringing' listen simultaneously on the cell phone and on the CUE channel of the console (fader TELECOM closed)

- To answer from DX822, make a short touch on the button "BLUETOOTH" (or you can answer from the cell phone screen).

Once the call is answer, the operator can talk with the caller:

- With the main fader closed (CUE zone) the caller listens on CUE circuit. Under this condition the "PGM" send to the cell phone is muted.
- Press TALK [6] to dialog with the caller. It will light in red. Press TALK again to end the conversation.
- Move the main fader to the zone "HOLD". The caller will hear the PGM signal, but is not on-air.
- To put the calling on-air, raise the main fader (TELECOM) until reach the appropriate level on VU meters.

#### NOTICE

Due to technical specificity of the digital communication systems, when a conference involves lines and cell phone or VoIP calling (Skype), the caller who is talking from the land line can hear an echo of their own voice. This echo is not on the air.

#### 3.2.4.2 END THE CALL

Make a short touch on the button "BLUETOOTH" [2]. Or hang out from from the own cell phone.

#### 3.2.4.3 Resume a call on the cell phone

To resume the call on cell phone; using the phone menu leave the mode "Headset" (or mode "Bluetooth") to switch the phone to the "speaker" mode (Speaker).

To return the call to the console, return to the "Headset" mode. This operation differs in each cell phone. Please consult the user's manual of the mobile phone.

Next shows a screen-shoot of a cell phone with Android, with Bluetooth mode enabled. The example shows the "Headset" icon, which transfers the call to Bluetooth device when it is power on; and return it to the phone when Headset mode is off. Others cell phones can display two buttons; a button with drop-down options; or other combinations. Refer to the cell phone user's manual.

#### 3.2.4.4 Make a call from cell phone

- In the cell phone, switch from mode "Bluetooth" to mode "Headset" as was explained in "3.2.2.3 - Resume a call on cell phone."

- Dial the number in the cell phone and call (SEND).
- To transfer the call to the console, return to the Bluetooth mode on the cell phone (see "3.2.2.3 – Resume a call on cell phone").
- Move the main fader TELECOM [9] to the zone HOLD. The cell phone receives the program signal, but remains off-air.
- To put the calling on the air, turn "AIR LEVEL" to the correct volume.
- To end the call from the console, short touch the button "BLUETOOTH". The call can also be ended from the mobile phone screen.

#### NOTES

The calling can be done from cell phone without leave the Bluetooth mode, using the console's talkback circuit.

- The ring tone and the voice of the caller listens on CUE circuit (main fader closed in zone CUE).
- To dialogue, use the button TALK [8] of TELECOM channel.
- To HOLD the call, move the main fader to the zone HOLD.
- To put the calling on-air, raise the fader to on-air zone
- To end the call, make a short tap on button "BLUETOOTH"; or hang-up from the cell phone.

### 3.2.5 Null factor

This adjustment is only for LAND lines. The rejection factor expresses the capacity of the hybrid to avoid that the transmitted signal returns distorted to the console. If the rejection factor is not well adjusted, it can produce coloration of the local voices from Studio when a land line is on air. Please refer to "2.6.1 – Null factor" to know how adjust the rejection factor.

### 3.2.6 Vox-IP communication (Skype)

Although the hybrid channel is not in charge of provides communications via Internet, we will explain here the procedure for the communications using a VoIP software (like Skype or similar).

#### REMEMBER

**USB 7 CANNOT BE USED IF CHANNEL USB 6 IS NOT CONNECTED;** since USB 7 is slave of USB 6. Both channels can be connected to the same computer.

WHEN TWO PCs ARE CONNECTED; CONSIDER THAT CHANNEL 7 CANNOT BE USED IF THE COMPUTER CONNECTED TO CHANNEL 6 IS OFF. Usually channel 6 is used with the on-machine; which is operative 24 Hs.

#### 3.2.6.1 Connection via Skype, or similar, with a remote journalist

To make this, proceed:

- a) Assign the microphones to PGM and REC.
- b) In TELECOM Channel move the switch PGM-REC to PGM.

Configure the VoIP software to use the audio devices that correspond to the Channel USB-2 of DX816/822 (typically Windows install as *USB2 audio device*)

Make a calling with Skype or a similar software using the computer that connects to the console's channel USB-2.

During the call, the signal from the Studio microphones sends to the remote location via Skype (using the bus REC). Note that when local journalists speaking, the VU meters on PGM and REC reacts at the same time.

The audio from remote journalist incomes to DX816/822 via USB-2. Raising the fader of USB-2 the Internet audio is on the air, so the local speaker and remote caller are communicated.

#### 3.2.6.2 Make a conference between Skype (or similar), land lines and cell phone

The Solidyne DX816/822 allows maximum flexibility to generate a virtual forum between journalists and interviewees, very far apart. Proceed as follows:

1. Keep the controls like previous description a) b).
2. At TELECOM channel move the switch PGM-REC to PGM.

Once communication is established via Skype (or similar) you will be in the previous situation. You may receive or generate phone calls from landlines or a cell phone. Under this conditions, the callers are sent to the air as usual, raising the fader of Telecom Channel. Now, turning on VoIP at TELECOM channel, a return signal from the Hybrid sends to Skype (through the REC bus). This will see reflected in the REC VU-meter, since it will shows the TELECOM signal. This way everyone are in conference.

#### WARNING

**DON'T SEND BACKGROUND MUSIC TO SKYPE. ONLY THE MICROPHONES MUST BE ASSIGNED TO REC BUS.**

Communication via SKYPE has a priority system that attenuates the speaker when he is interrupted. This makes it impossible to send the background music in the return signal, it will cause interruptions in the audio who's talking.

Due to technical specificity of the digital telecommunication, when a conference involves lines and cell phone or VoIP calling (Skype), the remote caller can hear an echo of their own voice. This echo is not on the air.

### 3.3 Monitoring panel

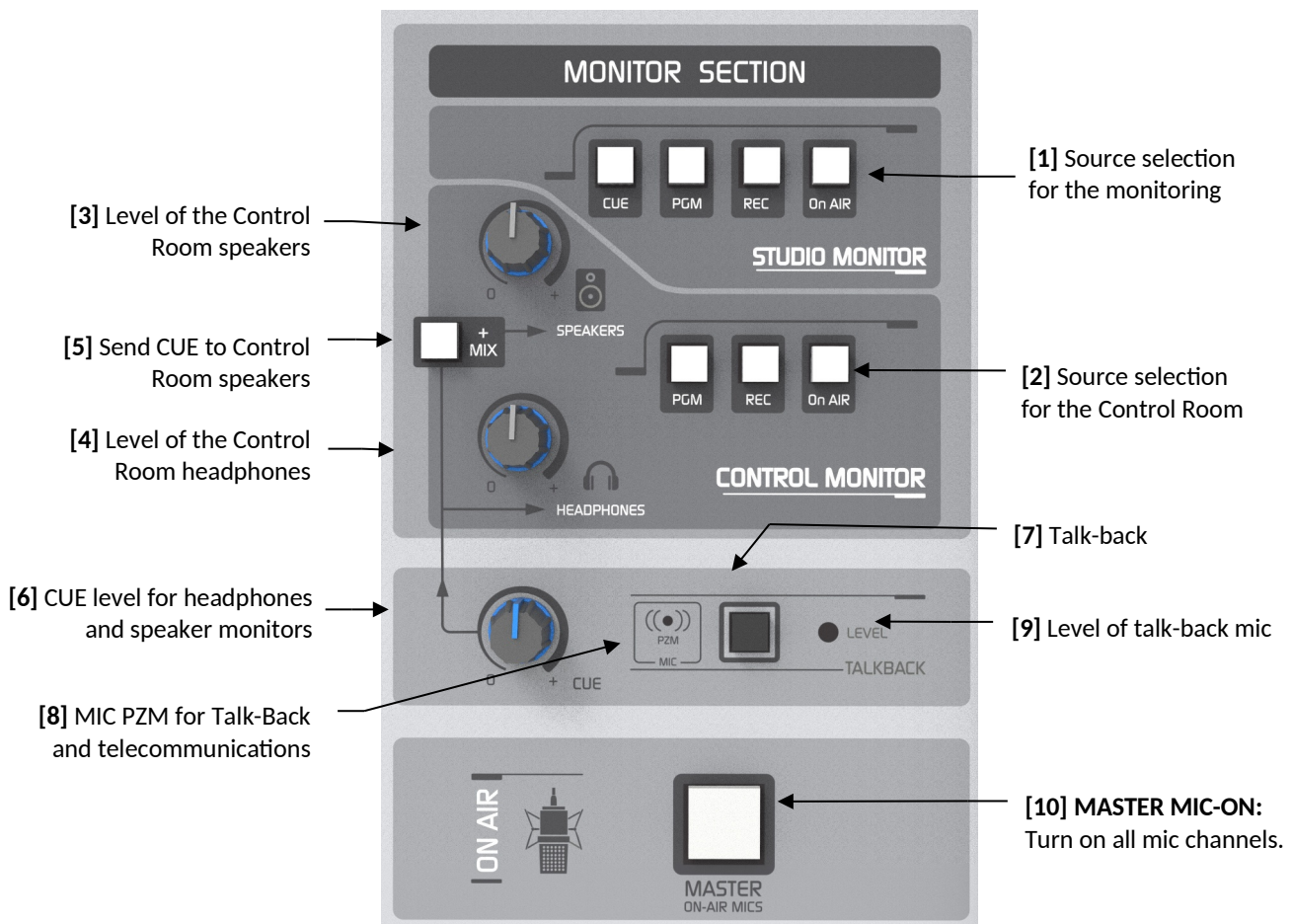


Figure 11: – Monitoring and talk-back

#### 3.3.1 Loudspeakers and headphones

The monitoring section shows the following areas:

- **Studio Monitor:** selects the signal for the headphones and loudspeakers of the Studio.
- **Control Room Monitor:** selects the signal for headphones and loudspeakers of the CR.

The section **STUDIO MONITOR** has 4 signal sources: CUE, PGM, REC and ON-AIR [1].

The DX816/822 sends to the Studio signal with fixed level, to headphones and loudspeakers. To manage the signals into the Studio, a headphones amplifier unit must be added. Solidyne offers the modules HD3 and HD5. HD5 manages up to 5 headphones with two monitoring buses (double input). HD5 has an output to the Studio speakers monitors, on-air counter and talk-back for communication to the Control Room (refers to “4 – Monitoring modules Solidyne HD3 & HD5”).

The section **CONTROL ROOM MONITOR** has knobs for the volume of headphones [4] and loudspeakers [3].



Using the own loudspeakers level's control, set the volume of the loudspeakers to a comfortable listening, leaving the knob SPEAKERS of DX822 at the center position.

The switcher [2] allows to choose the signal that will be listen at the Control Room.

The options are:

**PGM** to listen the output PGM. **REC** to listen the output REC. **ON-AIR** is for listen an external source, usually a tuner to hear the transmission from the air. Take in mind that the on-air signal is highly processed, for this reason the monitoring of the PGM signal is made from the air in the most cases.



If the Studio to Transmitter Link or the audio processor have a noticeable latency, the tuned signal can not be used to monitoring due to delays. In this case, an additional 3-bands audio processor can be used to generate a non-delayed processed PGM signal, that will be used only for monitoring.



### 3.3.2 Previous listening (CUE)

Each channel has a button “CUE” that allows to listen the signal when that channel is off-air. If CUE is pressed in several channels, the signals are mixed.

**At the Control Room, the CUE signal hears in headphones, and optionally can be sent to the main monitors.**

The knob **CUE [6]** set the level of CUE on headphones and, if was assigned, on loudspeakers monitors. The level of CUE is only managed by this knob, and is fully independent from others knobs like “Headphones” and “Speakers”.

To listen the CUE at the main monitors of the Control Room, press the button **MIX [5]**. The volume controls only with the knob CUE (the knob “Speakers” do not affect the CUE signal).



If the knob CUE [6] is closed, signal CUE is not listened although the knobs SPEAKERS [3] and HEADPHONES [4] be opened.

If the knobs SPEAKERS [3] and HEADPHONES [4] are closed, CUE will be listened on headphones (and speakers) if the knob CUE is enabled.

The bus CUE is monoaural.

#### NOTICE:

**The signal CUE will mix with others signals assigned to headphones (and speakers if the button MIX is pressed).** To listen CUE with detail, the operator can turn down the knob SPEAKERS or HEADPHONES according the case. To listen only the CUE signal, the operator can release the buttons PGM, REC and ON-AIR. We recommend to listen the signal on-air with a moderate volume and adjust CUE with a higher level, in order that both signals can differentiate.

### 3.3.3 Talk-back

The talk-back circuit allows the operator to dialog with those who are inside the Studio. To make this, the operator press the Talk-Back button. The Control Room monitors are muted, to avoid feedbacks. The operator’s voice will be listened inside the Studio, by the left channel. The right channel stays with the on-air signal, so that speakers and journalists do not lose the on-air reference. At the same time talk-back microphone will be listen on all studio headphones.

The operator can press CUE in a microphone channel to dialog with the Studio.



The level of the talk-back microphone is calibrated at factory, and usually it is not necessary to change it. If you need to change it, there is a pre-set called “LEVEL” to make the adjustment.

## 3.4 Microphones channels

### 3.4.1 Features

The Solidyne DX816 has 3 mic channels (faders), but allows to connect up to 5 microphones at the rear panel.

The model DX822 has 6 mic channels, with 10 microphone inputs (4 dual inputs). See “2.3.3 - Microphone inputs” for details.

All main inputs are 48V phantom powered (second “b” inputs are not powered) The voltage enables/disables for the three inputs at once.

The input gain adjusts with a preset using a screwdriver. The gain must be settled in order to the main fader works over the indication “OVL” (-30 dB).

The microphones has a global assignment to the buses PGM and/or REC (groups of three in DX822).

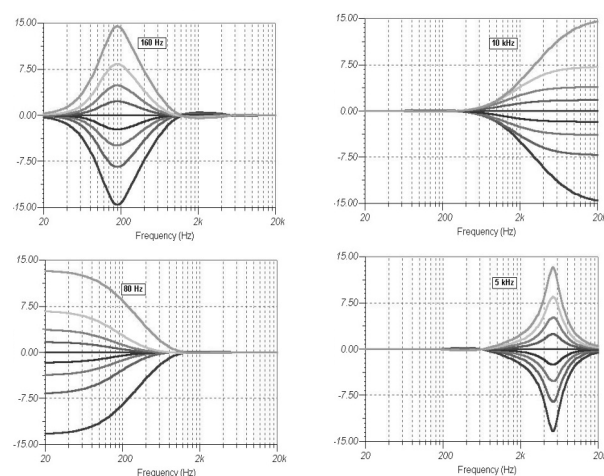
### 3.4.2 Processing

#### 3.4.2.1 Dynamic compression

The dynamic compression is applied only to microphone channels, to minimize differences of level between mikes. It starts to work when the peaks are over +2VU. Below this threshold the signal is not affected. The compressor ratio is 10:1, using fast attack and recovery times. The indication “MIC COMPRESSOR” shows the gain reduction applied to the signal in dB.

#### 3.4.2.2 EQ

The equalization settings are the same for all mic channels. The equalizer can be turned on/off with the button EQ. The GLOBAL EQ can be used to improve the sound of the voices; boosting or attenuating certain frequencies. Next figure shows the curves.



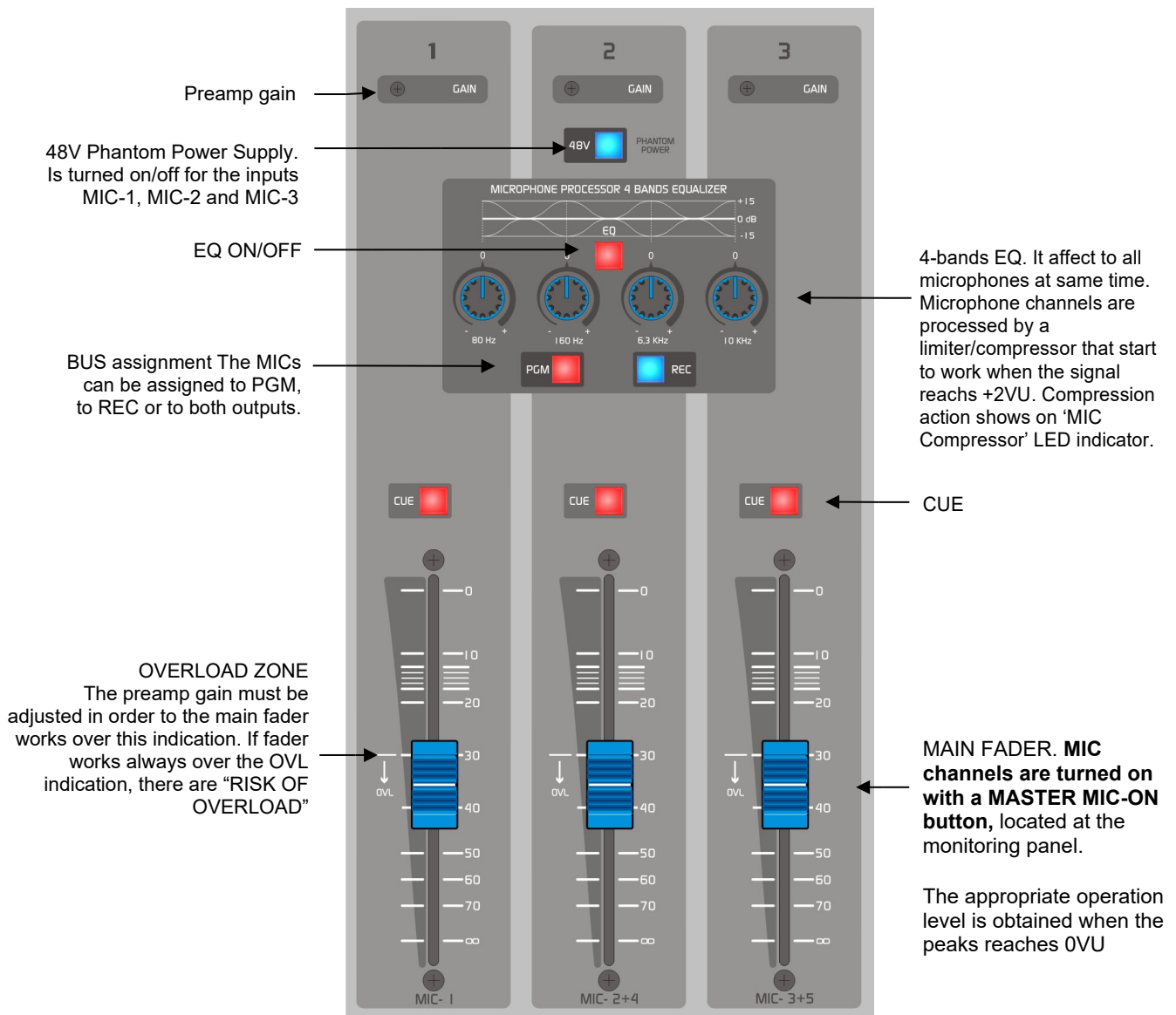


These curves are designed to improve the quality of the human voice. The ends of the spectrum are shelving type, with 15 dB of action.

As brief reference we said that the knob LOW (or bass) gives “body” and “weight” to the voices; whereas the knob HIGH adds “presence” and usually improves the intelligibility.

The central frequencies have a bell-shaped curve (peak EQ). Mid-bass is centered in 160 Hertz, where usually the popping effect are higher, or there is excessive resonance in certain male voices.

The mid-high band is located around 5 KHz, because around this frequency usually appears the problems with the sibilants.



### 3.5 Line channels

The Solidyne DX822 has four stereo LINE channels. Line channels have two stereo inputs. Channels 4 & 5 have balanced stereo inputs (RJ45 connector) and unbalanced stereo inputs (RCA), commutable from the frontal panel.

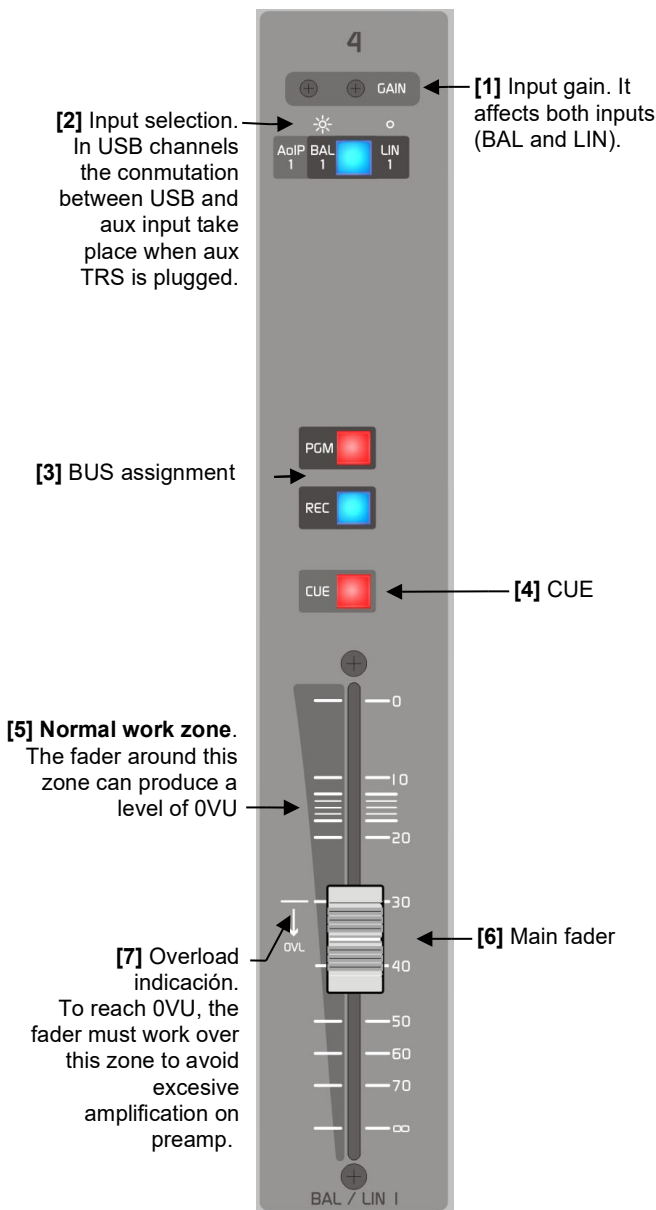
Channels 6 and 7 have digital USB inputs, and stereo unbalanced aux inputs with 1/8" Jack TRS. When the jack is plugged, the channel switch to aux input and the USB audio signal disconnects.

Line channels have gain adjustment that must be settled to obtain 0VU with the fader working above the OVL zone (the gray zone between -10 /-20 dB) to avoid excessive amplification on preamps. *NOTE: please see “2.2.2.2 - Digital inputs/outputs” and read carefully the note about the USB audio levels in Windows©.*

To put a channel on-air; assign the channel to PGM and raise the fader. The appropriate level is obtained when the peaks of the signal reaches 0VU.

Connections was detailed at “Chapter 2 – Installation and wiring”

Main faders of all channels work with ETM-VCA technology (Electrometric-Voltage Controlled Amplifier). The fader only manages DC, that controls an amplifier of variable gain. This method guarantees very low distortion and eliminates the noise by dust faders. Complete information about the new ceramic fader technology can be found at [www.SolidynePro.com](http://www.SolidynePro.com).



### 3.6 How to record

There are two stereo output channels. The main channel is Program (PGM) and is used for ON-AIR signal. Usually, the bus REC is used for recordings. The operator can make recordings while, simultaneously, the console is on the air.

### 3.6.1 USB recordings

The main output (PGM) and the recording output (REC) are available in digital format, through connections USB. When USB connects, Windows© will recognize recording devices “USB Recording device USB 1” corresponds to Channel USB-1 and gives the PGM signal. “USB recording device USB 2” corresponds to Channel USB-2 and gives REC signal (check this in Windows Control Panel/Audio and Sound Devices).

**On the computer:** configure the recording software to use Device USB-2 as recording device.

**At the console:** route the channels that wants to record to REC bus. The others channels remain assigned to PGM. Proceed as follows:

- a) In Monitor Section assign the Control Room monitors to the bus REC. To send REC to the Studio, assign REC for Studio monitor.
- b) Release PGM and assign to REC the channels that wants to record (i.e. MIC-1 and Line-4).
- c) Meanwhile the radio still on air through, for example, the channel 5 assigned to PGM with a musical program.

Now start the recording on the PC. To enable a MIC, remember to press the button Master MIC-ON.

The recording level is visualized on the REC VUmeter in the console and at the recording software. The input level is fixed for Windows mixer. To adjust the recording level, use the DX816 channel faders. Peaks must reach 0 VU (0Vu = -15 dBfs).

**RECORDING DEVICES DO NOT HAVE GAIN CONTROL BY SOFTWARE**

At the Studio the speaker can monitoring the signal using headphones or at loudspeakers. When the recording ends, it can be checked using the same REC bus, routing to REC the channel in which the PC playing. At this point; be careful of not producing a feedback loop in the computer. make sure to mute in the pc (from windows mixer) all sources of signal except “wave” and “gain control” (main).

When the work is finished, return all the controls to the original position, to leave the console to the normal way of operation.

### 3.6.2 Record a call

The DX816/822 allows routing the TELECOM channel to PGM or to REC. It allows to record calls while a musical program is on air. For make this, assign the TELCO channel and a microphone to REC; and proceed with the communication as was explained in “3.2 - Hybrid & Telco”. Assign the monitoring to REC to listen to the communication (see previous issue).



## 4. Solidyne HD3 & HD5 monitoring docks

### 4.1 Features

The optional accessories Solidyne HD3 and HD5 are compact docks that concentrates all connections needed to the monitoring at the Studio.

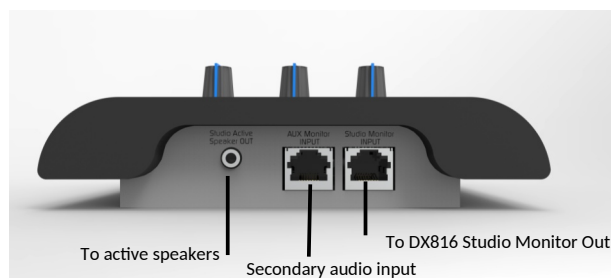


The main features are:

- **Connection for up to 5 headphones** in HD5 (3 in HD3) with independent level knobs.
- **Double bus for monitoring from two sources.**
- **Output for active speakers**, with gain control.
- **Tally-light.**
- **Talk-back** into the Studio. Allows to talk with the Control Room.
- **Timer/Clock** (only in HD5)
- DX816/822 supports up to three HD3 or two HD5 docks, connected in cascade.

### 4.2 Connections

The monitoring dock Solidyne HD3 connects to the console using a standard **STP** (Shielded Twisted Pair) cable, with **RJ45** at both ends (pin to pin). This cable is provided by Solidyne. The cable connects the “Studio monitor OUT” of DX816/822 (See Figure 6, [7]) with “Studio Monitor Input” at HD3.



The headphones outputs uses Jacks TRS 1/4". Headphones of 16 and 32 ohms can be mixed.

The output for active speakers uses Jack TRS 1/8" (minijack).

**AUX input:** Allows you to enter a second source for monitoring in Studio. The headphones can be switched between the main signal (selected in the console, usually a PGM air signal) and the secondary audio source.

The input is unbalanced, with RJ45 connector. The connection to an audio output on RJ45, balanced or unbalanced, is done using a standard cable STP, RJ45 at both ends.

The connection of the auxiliary input is as follows:

#### UNBALANCED INPUT (AUX)

PIN RJ45	CABLE COLOURS
1 Left channel (+)	Orange / White
2 NC	Orange
3 Right channel (+)	Green / White
4 GND	Blue
5 Reserved	Blue / White
6 NC	Green
7 -15 (option)	Brown / White
8 +15 (option)	Brown

### 4.3 Using the monitoring docks

#### About the headphones at the Studio


The dock HD5 can manage up to 5 headphones, with independent level knobs.

HD3 manages 3 headphones. Has two level knobs. The headphones 2 and 3 shares the same knob.

The console gives a signal with fixed level to the monitoring docks. The signal is switched from the console (PGM, REC, ON-AIR or CUE).

#### About the speakers monitors at the Studio

The output “LOUDSPEAKERS” has their own level knob. The console gives a signal with fixed level to the monitoring docks. The signal is switched from the console (PGM, REC, ON-AIR or CUE).

 Using the own loudspeakers level's control, set the volume of the loudspeakers to a comfortable listening, leaving the knob SPEAKERS at the center position.

#### About the INTERCOM in HD3/HD5

The white button has double functions:

1. When the microphones are off-air, press this button to talk to the Control Room. HD3/HD5 has a built-in microphone.  
At Control Room, the operator listen to the HD5/HD3 at the speakers. To answer, the operator uses the conventional talk-back circuit as was detailed “3.3.4 – Talk-back”
2. **When the Studio is on-air, the white button lights red**, to indicate that the microphones are on-air. Under this condition, the talk-back disables.



The *preset* located under the mic adjust the gain of the dock's microphone.

### Counter/Clock (only for HD5)

The display shows the current time (when mics are off-air) or the time lapsed on air (when microphones

are on air). The "counter mode" can be disabled; quitting an internal jumper. Open the back cover of the Studio Box and to retire the jumper. There is a unique jumper available. This will always deactivate the on-air lapsed time counter, being always the time on display.

## 5. Maintenance

In order to obtain great results with DX816/822 consoles, we recommend to read the following recommendations.

### 5.1 How to get a long life of the console

Maintain the room clean and free of dust. The surface of the console can be cleaning using a very smooth detergent (the kind used for painted walls) and a fine cloth hardly humid. NEVER USE alcohol, benzene or petroleum derivatives.

Take this rule: NO SMOKING at the control room. The cigarette ashes are LETHAL for the faders and switches. It affects, in addition, to other equipment of the radio (CD players, microphones, etc.). By the same reason, don't drink or eat near the console.

DX816/822 consoles use the new Conductive Ceramic fader technology with more than ten million operation guarantee (usually 15 years of use). Please see more information at [www.solidynePRO.com](http://www.solidynePRO.com)

This faders do not need maintenance nor cleaning.

### 5.2 Preventive Maintenance

The DX816/822 console is manufactured using hi-tech integrated circuits and heavy-duty electronic components, that guarantee an excellent reliability and allows eliminating the routines of preventive maintenance.

Usually the console do not need special maintenance routines. Be sure the operator's has a gentle use of the console. Remember that the rear connectors are not designed for daily use; avoid connection of headphones directly to console. When the operator's changes frequently its headphones, use an external patch panel to avoid wearing console jacks

### 5.3 Spare parts

All faders are assembled using wired connectors, and they are mounted to the chassis with two screws, so its replacement is simple.

Once outside, any module is easily repaired thanks to that all the components are clearly identified in the Service Manual (see 5.4). Usually we recommend to acquire the complete module and to change it because its low cost does not justify to repair it.

### 5.4 Service Manual

Service manuals, with the electrical circuits and components layout, are sent by email to authorized Solidyne dealers and radio stations who subscribe a confidentiality agreement. This Agreement will be sent by FAX to Solidyne (5411 4702-2375) in paper letterhead of the radio and signed by its Director or General Manager. The information is encrypted.

For details please visit [www.SolidynePro.com](http://www.SolidynePro.com) Solidyne brings full support to technicians during the repair work. Please contact us at: [info@solidynePRO.com](mailto:info@solidynePRO.com)



# 6. Technical specifications

## AUDIO INPUTS

DX816: 5 balanced MIC inputs (XLR) [Fader1=MIC-1; Fader2=MIC-2+MIC-2b; Fader3=MIC-3+MIC-3b]  
DX822: 10 balanced MIC inputs (XLR) [Fader 1=MIC1; Fader 2=MIC2+MIC2b; Fader 3=MIC3+MIC3b; Fader 4=MIC4; Fader 5=MIC5+ MIC5b; Fader 6 = MIC6+MIC6b]  
2 balanced stereo line inputs (RJ-45)  
4 unbalanced stereo line inputs (2 RCA - 2 TRS 1/8")  
1 external stereo input (TRS)  
2 digital stereo inputs (USB)

## INPUT LEVELS / IMPEDANCE

MIC= -10 dBu/-75 dBu; 150/250 Ohms  
Balanced lines = -20 dBu/+26 dBu; 600 ~10Kohms  
Unbalanced lines = -15 dBu/+12 dBu; 600 ~10Kohms

## PHANTOM POWER SUPPLY

48V ON MIC-1, MIC-2 and MIC-3

## ANALOG OUTPUTS

2 stereo outputs; PGM & REC, balanced +4 dBm;  
Max level +28dBu (10K), +20dBm (600 Ohms)  
External hybrid: unbalanced +4dBu

## OUTPUT LEVELS / DIGITAL OUT

Agrees with K-15 recommendation: 0VU = -15 dBFS

## MONITORING OUTPUTS AND HYBRID

1 stereo output to Studio (RJ45). Level +4dBu for active loudspeakers. Are muted with mics on air.  
1 stereo output for Control Room (mini-jack). Level +4dBu for active loudspeakers.  
1 stereo output for Studio headphones (RJ45). Level +10 dBu  
1 stereo output for Control Room headphones. +10 dBu  
1 send for external hybrid (MIX-Minus) +4 dBu/10K

## CUE

CUE bus (mono), switchable to loudspeakers and/or headphones of Control Room and Studio loudspeakers.

## HEADROOM

22 dB @ LIN to PGM Ref + 4dBu/10 k

## GPO

At MIC-1, MIC-2 & MIC-3  
For lines 6 & 7  
Open collector +24V @ 0,1 A

## FREQUENCY RESPONSES

20-20.000 Hz +/- 0.5 dB (LIN to PGM)

## MIC EQs

4 bands. 80 Hz top flat curve, 160 Hz bell shaped, 6,3 KHz bell shaped, 8 kHz top flat. Action +/- 15 dB

## DYNAMIC COMPRESSOR

MIC compressor of low distortion, of automatic action. Threshold = 0VU. Attack < 10 ms. Ratio: < 2 dB variation for a increase of 15dB of the mic signal. 20 dB max compression

## NOISE

MIC in, EIN=-120 dBu/150 ohms  
LINE in, S/N > 75 dBA

## DYNAMIC RANGE

From Linea to PGM >90 dBA

## CROSSTALK

PGM-REC > 70 dBA @ 1 kHz

## DISTORTION

From LINEA to PGM < 0.03 % THD @30-15.000 Hz.

## PHASE

From line to PGM, < 3° L&R @50-10.000 Hz

## STEREO TRACKING

Error below 0.2 dB L/R in rage from 0 to 40 dB of fader

## TELEPHONIC HYBRID

Active hybrid of 3 lines (2 land lines + cell phone by Bluetooth 4.0 whit sampling at 16 KHz). Include a mini-PBX.  
Frequency response: 300 - 3.400 Hz  
Noise: > 60 dBA S/N  
Rejection: > 40 dB - Rejection adjusts from the frontal panel  
Priority system: 12 dB when the local speaker interrupts  
Logic of automation: Audio & Logic are managed from a single 100mm slide fader that performs all the operations in error-free mode:  
CUE (closed fader) with "hands free" circuit  
Hold with PGM return to the caller.  
Onair: Level of the calling on the air.

## PROTECTION AGAINST LIGHTING

Audio & Logic are managed from a single 100mm slide fader that performs all the operations in error-free mode (CUE, private hands free Talk - Hold with Air return - Live On Air)

## TALK-BACK

Included Talkback MIC, with Audio Limiter. Noise Canceled PZM type. Outputs to Phone Line Hybrid or Studio Speakers & Headphones.

## VU-METERS

4 electronic VU-meters por PGM & REC. Cuasi peak reading, with standard yellow VU scale. Microphone compressor indicator 0 - 15 dB

## TALLY LIGHT

ON-AIR signal output (turns on when MIC is open). Tubular connector 2.1 mm (internal pin). 12 V CC @ 0,15 amp (two Solidyne On-Air lights).

## POWER SUPPLY

Tubular connector 1.8 mm (internal pin).  
Switching external power supply 90-240 V, 20VA [28V@0,57A]

## DIMENSIONS AND WEIGHT

Depth: 350 mm; Wide: 550 mm Height: 130 mm  
Shipping weight: 8 Kg  
Packaging (mm): 600 x 500 x 180