SOLIDYNE
DIGITAL POWER

D612
broadcast console

OWNER'S MANUAL

www.solidynepro.com

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1.1 About this manual

REVISION, December 2022

1.2   Whats in the box

When unpacking the unit; check the contents to verify that the console has not received blows during the shipment. Inside the packaging you receive the following elements:

- *1 switching power source 90-240 V*
- *This user manual*
- *1 guaranty agreement*
- *1 tubular connector (2,5 mm) for Tally Light*
- *4 self-adhesive rubber pads.*

Optionally:

- *Wiring kit (optional). If you did not acquire the wiring kit, you will receive the corresponding DIN connectors.*
- *One RJ45 Ethernet wire (only models /AoIP)*

1.3   Features

D612 console is a compact, solid and elegant unit; with a great flexibility for interconnection. It includes USB digital I/O which brings digital link with the computer; avoiding the use of sound cards in the computer.

D612 has 5 100mm professional Conductive-Ceramic fader of with ETM-VCA control that offer a life utility of one million operations (over 15 years). It is the best solution for small and medium radios; and for auxiliary recording studies in great stations.

1.3.1   Input channels

Two microphone channels manages four microphone inputs. Each mic channel has dual input.

All inputs supports 48V phantom power.

Microphone channels has a processing stage that includes **4 bands equalizer** and automatic dynamic range **compressor**. Each channel can be sent individually to the air signal (PGM) or as-signed to the processing stage. MIC-3 channel can be switched to AUX stereo line input.

Each mic fader can manage two dynamic microphones. At the rear panel you find 4 XLR inputs (MIC-1a; MIC-1b; MIC-2a; MIC-2b). The built-in dynamic compressor compensates the level between the microphones. It's appropriated that mic's connected to the same channel are the same model.

3 LINE channels allow connecting up to 6 sources. Channels 3 and 4 manages unbalanced analogical inputs (Line-1 and Line-2) and balanced inputs (Bal-1 and Bal-2) whereas channel 5 are digital, with direct connection to the computer via USB. In addition, all channels have one analog auxiliary input.

All the channels have a gain control, that take effects on the selected input (main or auxiliary).

✎ In units with the /AoIP option, the LIN-2 auxiliary input is internally assigned to the audio signal of the streaming module, and cannot be used for other signals.

"Telephone Hybrid" manages one land telephone lines (2 wires) and one cellular phone linked via Bluetooth. Both can be in conference. The use is very simple, since many functions are automated. Additionally, the console has send and return connector for external hybrids.

1.3.2   Monitoring

The monitoring area routes the signals to the loudspeakers and headphones for the studio and the control room.

Studio & Control Room

Has a unique level knob for loudspeakers and headphones; and the assignment switches that allow select between ON-AIR signal (external tuner), console output (PGM) or recording output (REC). Usually the level in the console is fixed; since a headphones mixer allows to manage each headphone and active speakers has it own level knob.

CUE monitor

The console has a built-in loudspeaker for cueing, with its own control level. CUE is an internal send that allows listening to the audio signal previous to the fader (PFL).

When D612 is used into a "hot studio" (mics and mixer located into a unique room) the built-in

speaker can be disabled and the CUE signal sent to the headphones.

In addition, the monitoring section contains the Talkback circuit; that works in combination with CUE to make a private dialog from the hybrid with a calling.

1.3.3 Outputs

The Solidyne D612 has **two main outputs** named PGM (Program) and REC (recording). Phone lines are sent only to PGM.

In addition it has an **USB digital output** that appear in the computer as *external sound recording devices* (USB recording devices) and sends the program signals.

The outputs are:

1. **PGM,** The **PROGRAM** output is used the send the signal to the air.

2. **REC**, used to recordings (and allows to listen a channel in the main monitors without send it to the air).

3. **PGM/REC USB,** program or recording output is also available via USB.

4. **PGM over Ethernet (optional):** Available in models with the expansion /IP. The console can send RTP streaming to link with the transmitter plant; or mpeg streaming to upload to an Icecast server (see Chapter 4 – Models D612/IP)

D612 has send & return connections for **external hybrid**. The audio from the external hybrid enters trough the "hybrid channel" and operates just as the built-in hybrids; which continue being operative. Please note that land lines of internal & external hybrids will work in conference.

The **Tally output** allows the direct connection of ON-AIR LED's lights. Tally activates when opening the mic channels. The mixer has built-in tally LED.

TALLY OUTPUT LOAD DOES NOT HAVE TO EXCEED 120mA (two Solidyne ON-AIR lights).
If your radio uses the old incandescent light bulbs, a 12 V relay must be connected.

1.3.4 ETM – VCA faders

Using a technique of control called ETM-VCA (Electrometric Voltage Controlled Amplifier) the faders DO NOT manage audio signals. The change on the level is made using low noise amplifiers of great stability.

The main faders manages control signals that changes the gain of the electronic amplifiers. For more information please go to solidynepro.com

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 two million guaranteed operations (about 15 years of use).

2.1 Overview

Installation of Solidyne D612 doesn't present particular jobs. However, keep in mind the basic rules for all professional audio installations. The balanced inputs of the D612 uses DIN-5 multipin connectors (provided with the console). In option, you can acquire the Solidyne wiring kit (MNG612), which include all the cables and connectors needed for the set up.

For console grounding use the rear side **GND bronze tip** with a 2mm cable to a good buried cooper bar.

2.1.1 Parasitic signals

All unwanted signals that appear in audio lines usually consider *parasitic signals*. A common type are denominated *humming*, low frequency signals (multiples of 50/60 Hertz) caused by the interaction of electromagnetic fields coming from the AC line. When the interference source is a magnetic field (generally originated in a supply transformer) the resultant interference will be denominated *electromagnetic humming*. When the interference is due to such electric potentials as cables that take supply tensions, that are elevated in comparison with the audio signals present in the circuit, you will be in presence of *electrostatic humming*. The distinction is not merely academic, because the resolution of a problem supposes the knowledge of the noise type to apply the correct solution.

Examples: To minimize the reception of electromagnetic humming in the wires, remember the following rule: "THE AREA AMONG TWO AUDIO WIRES WILL BE MINIMUM." It implies that the cables will be tied very close, like the shielded twisted pair audio cables. They should pass far away from any transformer or devices that manage high-intensity currents. Is important to remember that a wire can be good shielded, but if his conductors don't complete the conditions of minimum area it will be susceptible to take magnetic humming.

Other parasitic signals are: AC HUM, RADIO FREQUENCY and CROSSTALK. As hum noises like the radio frequency are originated by electromagnetic fields of high frequency; the first ones are originated by disturbances due to the connection and disconnection of equipment's to the AC line, the seconds ones are generated by communications transmitters or industrial equipment. If these signals penetrate into audio lines, with sufficient intensity, can surpass the action of the special protection filters, and to reach some sensible part of the input stages. In that case, the interfering signals can be demodulated and already turned audio signal, and will be amplified by the rest of the system. It is fundamental, therefore, to maintain the in-terference within reduced margins. It is obtained avoiding very long audio lines, with aerial sections or that pass near of transformers or RF transmitters. For protection against very high frequencies is advisable to use double shielded cables, guarantied by the manufacturer.

CROSSTALK is the reception of signals coming from other lines of audio. This, like all unwanted noise, it can be supposed controlled when its level is below the level of the system residual noise. Then, all considerations mentioned for the case of buzz are valid.

2.1.2 RF interference (Hum)

D612 consoles have numerous internal protections against RF fields, for the AM and FM broadcasting band. When the transmission station is installed correctly, there will be no interference problems, still with FM equipment of 50 KW installed in the terrace of the radio station. Nevertheless, when the antenna is badly positioned with respect to the Studio or has severe SWR problems, then it does not have the minimum value of field intensity, downward. Or maybe there is a faulty ground connection, then, strong standing waves will appear on the cables of the Studio that can induce high electrical currents inside the audio console.

Symptoms: If the interference takes place at the A.M. band, the sound transmitted by the A.M. radio will be listened in the loudspeakers on background (or at buses PGM, REC). In case of FM transmissions, the interference inside the console demodulates the A.M component of the FM carrier; (usually hum from power supply) causing background humming, because in many transmitters of FM, the final output stage is not powered with stabilized tension. Therefore, if console D612 presents humming, please make a test shutting down the transmitter a few seconds to check if the problem disappears. Sometimes, an FM transmitter with the output stage badly calibrated also produce interference in which you can hear the transmitted audio (due to the misalignment a part of the FM modulation is translated to AM modulation).

Solution: D612 consoles have internal filters against RF interference. Therefore, if interference's appears, normally is caused by installation problems on the transmission station that generate elevated currents in the wiring of audio on the Studies, due to the standing waves. These currents circulates inside the cabinet of the console, and interferes to the circuits when overpass the barriers that impose to this currents the built-in RF filters. The solution, then, <u>must be external to the console</u>.
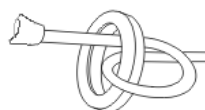
To solve it, pass all cables through **ferrite O-rings** of 2" diameter. They will be used in each connector, of input and outputs. The total set of cables that goes to each connector, will have to be twisted around the ring.

2.2 Power source

At the left side of the rear panel you will find the connector for the power supply. The switching power supply (provided with the unit) works with any AC network (90-240 V) and tolerate great variations of tension. Gives **regulated 28VCC.**

⚠️ **PLEASE DON'T REPLACE THE ORIGINAL POWER SUPPLY.**

The console doesn't have on/off switch. It remains fed while the power supply is connected.

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 Input channels

On the rear panel are located all the inputs and outputs connectors. Take in mind that good connections offer safe and free of faults operation. By this reason we recommend you: take it the time necessary to make the connections carefully and use always material of first quality.

2.3.1 LINE inputs

Each line channel has two stereo inputs, that selects from the frontal panel. **'BAL-1'** and **'BAL-2'** inputs are symmetrical balanced, using a DIN connectors. **'LIN-1'** and **'LIN-2'** inputs are unbalanced, and uses Phono-RCA connectors.

✍ In units with the /AoIP option, the LIN-2 auxiliary input is internally assigned to the audio signal of the streaming module, and cannot be used for other signals.

To connect an equipment with balanced output you will need a cable from DIN-5 to two CANNON (XLR) or stereo Plug's (TRS ¼") according to corresponds. The pin distribution of DIN-5 and XLR are described next. Remember that you need to connect TWO XLR to the DIN (balanced left and balanced right).

DIN 5 (stereo bal.)	Female XLR
1 AUDIO (-) RIGHT	**1** GND (shield)
2 GND (shield)	**2** AUDIO (+)
3 AUDIO (-) LEFT	**3** AUDIO (-)
4 AUDIO **(+)** RIGHT	
5 AUDIO **(+)** LEFT	

Unbalanced lines manages signals of -10dBV @ 10KOhms (home type or semi-pro).

The GAIN control is common for both inputs; that is to say, that operates on the selected inputs.

2.3.2 USB audio interface

The Channel-5 is for direct connection to the computer using an **USB** port.

Standard cable USB A/B are used. Ports 1.1 or 2.0 are supported. Place the CPU so next to the console as it will be possible, in order to use a wire of 2 to 3 meters of length. USB extension can be used; but it is recommended **not to exceed 4 meters.**

The channel must be connected to a computer running Windows©7/10; Linux or OSX. When connecting the channel to the PC, Windows© automatically recognizes it and installs the correspondent drivers. Additional drivers are not required.

⏚ Before connecting the USB cable, make sure the computer has effectively grounded through its terminal on the rear panel.

If you want to check the ground, connect a tester on the scale of 25 Vac between the chassis of the PC and console, and verify that the voltage is zero volts (+ / - 5V). Only then connect the USB.

If the equipment is not properly grounded, could damage the USB port on the console or computer.

Windows© detect the USB channel as **"USB audio device"**. At the console; the display indicator "Digital In/Out" will show "U4", indicating that stereo input and output of USB channel was recognized.

Remember that you can see the playing and recording devices from "Control Panel > Audio and sound Devices>Audio". In this window you define the preferred devices for Windows (those that applications use by default). The devices are selected from a drop-down menu.

Configure the USB devices in the automation and editing software. If the computer does not have installed a soundcard, default playing and recording device will be the USB device (Channel-5 input and PGM or REC output).

In addition to the input, USB offers a **digital send to the computer**, that appears in Windows© as **"USB recording device"** (PGM or REC). So that you have available in the computer, the program or recording signal.

⚠️ **ABOUT USB DETECTION**

Windows 7: Check if Windows has properly recognized the recording device. If Windows recognizes the device wrongly as "microphone device", the recording will be mono (same signal in both channels). To fix it, go to: *Control Pannel → Sound → Recording →* and choose *USB* device (appears as USB microphone). Press *[Properties]*. Then select *'Advanced Options'*, open the drop-down menu Recording format and choose stereo (2 channels, 16 bits, 44.1KHz).

2.3.3 MICROPHONE INPUTS

Microphone channels are balanced electronically, with XLR standard connectors.
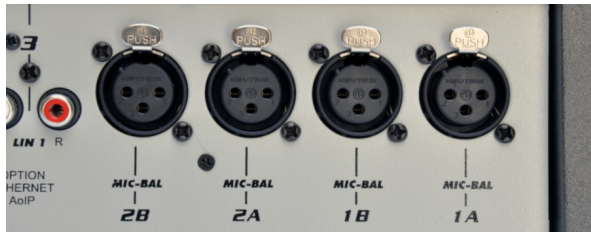
Each channel has two XLR connectors that are internally mixed. If connects two microphones to the same channel, use dynamic type, the same make and model. Level differences in the voices are balanced by the internal audio compressor. The action of the compressor depends on the level of the PGM signal and is indicated on the console.

The "48V" switch on the frontal panel enables the phantom power for all mics. Remember that dynamic microphones can receive 48V without risk of damage since its coil decoupling the DC.

2.4 AUDIO OUTPUTS

2.4.1 PROGRAM AND RECORDINGS

Program output (PGM) is the main output that takes the signal that will be emitted to the air. It is balanced, with female TRS connectors.

Recording output (REC) is balanced with Phono-RCA connectors.

In order to connect PGM outputs to an unbalance input, connect **only the (+)** pin, and leaving unconnected the (-) pin.

To connect the balanced TRS output to a balanced XLR input, the cable will be:

✍️ **Balanced connection Plug to male XLR** *(for each channel)*

TRS (Plug)	**XLR** (Cannon)
GND = sleeve	GND = pin 1
signal (+) = tip	signal (+) = pin 2
signal (-) = ring	signal (-) = pin 3

⚠️ Please do not connect pin 2 or pin 3 to ground. Never uses "mono plug' s" (TS) to connect this outputs.

The **nominal output level** is **+4 dBu @ 0VU**. If you unbalance the outputs, take in mind that the level reduces in 6 dB, therefore 0VU = -2dBm.

Note that if some nonprofessional equipment is connected to the balanced output, may the level be too high, causing saturation. In those cases a fixed attenuation must be used to reduce the level (a resistive splitter is a good solution).

2.4.2 USB send

USB allows direct connection with a computer, and gives PROGRAM or REC signal (it switches from "USB OUT" in Channel-5). This feature don't requires special drivers. Windows automatically recognizes the USB device (see "2.2.2.2 – USB).

Usually USB signal will be used for direct to hard disk recordings; or webcasting. For details of use the USB outputs in Windows© please see "3.6 - Recording of programs".

2.4.3 MONITORING OUTPUTS

2.4.3.1 HEADPHONES

There are outputs for STUDIO and Control Room headphones. A unique knob manages the level of the headphone and speakers outputs.

At the Studio, usually is used a headphone's mixer like the Solidyne Studio-box (not included) to distribute the signal to each people. See "Chapter 5 – Studio Box" for details.

2.4.3.2 MONITOR SPEAKERS

D612 was designed to work with active loudspeakers. The STUDIO and CONTROL outputs work with line level at 0 dBu (0,775 V RMS). They have independent controls for bus assignment and level. Both outputs use female 1/8" TRS connectors.

✎ Remember that studio monitors are muted when mics are on-air. Control Room speakers are muted when the button Talkback is pressed.

2.4.3.3 CUEING

By default, the CUE signal **is sent to the built- in speaker only**. When the mixer is used in a single room (aka "hot studio") with the microphones, the CUE speaker can be disabled and the CUE signal can be mixed to the operator´s headphones (Control Room headphones).

This setting requires to change an internal connection. The procedure is the following:

1. Flip the console to access the screws on the bottom. Rest the equipment on a soft surface to avoid damage to the paint on the sides. Remove the 6 screws indicated in the image.
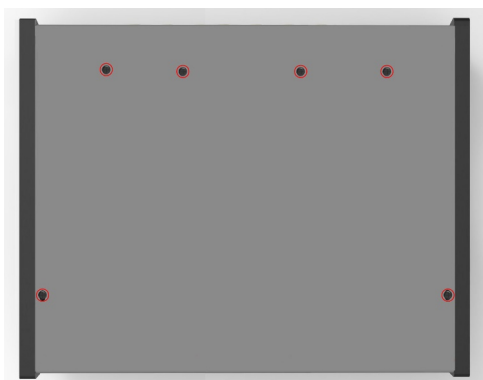
2. Unplug the wire from the speaker.

3. Plug the wire to the 2-pin connector indicated below:
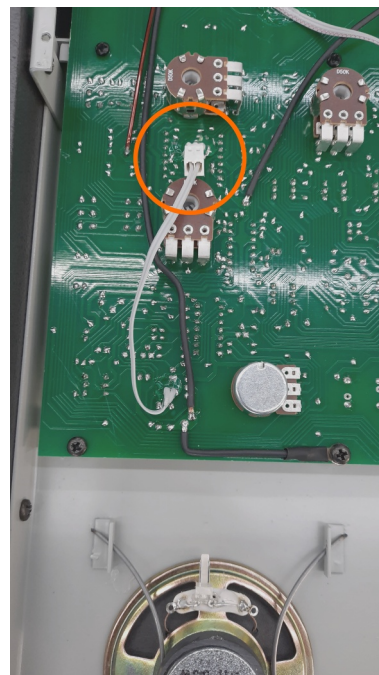
This connection send CUE signal to CTRL Room headphones

2.5 Tally light

Tally light output gives **12V/120 mA** when MICs are turned on from the Master Mic button. This output supports the connection of two standard Solidyne "on-air" tally lights.

NEVER exced 120 mA consumption on Tally output

The console has a built-in light; that resolves the on-air signalization for small studios.

2.6 Telephonic Hybrid

2.6.1 CONNECTING A STANDARD LINE

The hybrid channel manages ONE land telephone lines and ONE cellular phone. The internal adjustments are from factory; so the user do not needs to

make adjustments. Only the null-factor must be adjusted for the land line, only once on setup.

On the rear panel there are two RJ11 connectors; for the telephone lines and the associated phone set. The line can directly be connected to public central telephone or the local central (PBX) of the radio station. The private central sometimes deteriorates the hybrid rejection, reason why we advised to directly connect the console to the public line, whenever it is possible.

The telephone lines internally are protected against lightings by metallic oxide varistors (SIOV). Nevertheless, we recommend to use additionally external line protection of good quality.

> ⚡ Although the console has internal protection; it is recommended to use external line a protection against transient voltage peaks due to accidents or lightings.

2.6.1.1 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 air. Whichever greater it's this factor more "clean" will be the sound quality of the local speaker on the air. In order set the rejection, proceed as following:

1. Make a calling through one hybrid of the console. Ask a speaker to talk into the microphone to the interviewee. Increase the level from "Telephone Control" until the voice of the interviewee move the PGM VU meter into the red zone.

2. Listening to the voice of the local speaker on main monitors, turns the preset "Null" until reducing to zero the distortion superposed to the speaker's voice. Please make this adjust when the console is installed.

3. In case that the phone line connected to the console be changed, this adjustment must be repeated.

2.6.2 CONNECTING A CELL PHONE

It requires a phone with Bluetooth connectivity. Any phone with Bluetooth can be linked to Solidyne D612, eliminating the dependency on the adapter cable "hands-free". While the phone can be up to 10 meters away from the console, you should leave it near, but not on the console. The procedure to link the phone to the console is similar to that used for other Bluetooth devices.

By linking your cell phone to the console creates a link between the two devices, and allows the phone to store the ID of the console. This operation is performed only once (for one cell). Once the console and phone are connected, the console automatically connects to the phone when the Bluetooth link is enabled on both devices.

Procedure:

1. **At the console,** Make sure that the LED of "Cellular Phone Bluetooth" is off. When Bluetooth is on, the LED "Cellular Phone Bluetooth" flash in blue slowly (one flash every 2 seconds). In this case, turn off Bluetooth pressing and holding until the flashing button turn off.

1. **At the console,** enable the "discovery" mode. This mode only can be accessed if Bluettoth is turned off. Being the Bluetooth turned off, press and hold by 10 seconds the button "Cellular Phone Bluetooth", until the LED fast blinks in blue-red. It indicates that the console is in "Discovery" mode, to be recognized by by the cell phone.

2. **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 cellphone users manual.

3. When the searching ends, look for the D612 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 of the Solidyne D612 is now stored in the cell phone. The devices are "paired", no need to repeat this operation while using the same phone. The Bluetooth indicator light on the console switches to slow flashing, indicating "Bluetooth is 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 mobile

To reconnect a cell phone, proceed as follow:

1. Turn on the Bluetooth of the D612 by pressing "Cellular Phone Bluetooth" until the LED lights (5 secs aprox). The button will flash slowly in blue.

2. 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 Setup the level

VERY 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.** Usually, the volume of the phone's speaker is independent of the volume 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 the Bluetooth level.

To change the Bluetooth level, make a call using the console (or a hands free headset) and set the Bluetooth level to the maximum. This setting is stored at the cell phone's memory. **If you use another cell phone with the console, you needs to set the Bluetooth level again**.

2.6.2.3 TURN OFF THE BLUETOOTH

To turn-off Bluetooth at the console, press and hold "Cellular Phone Bluetooth" until LED off.

2.6.3 EXTERNAL HYBRID

D612 supports external hybrids using a ¼" TRS for **send and return signals**. "External Hybrid" sends PGM signal through the "tip" and receives through the "ring" (return or input to console).
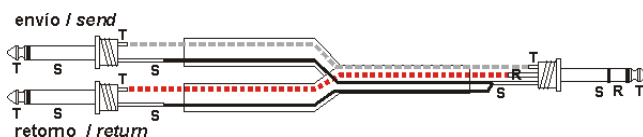
Ext-Hyb output (Tip); Is the return from microphones from the console and PGM to the phone lines. This is a MIX-MINUS signal, and connects to the input "return to phone lines" at the external hybrid (audio input of the hybrid).

Ext-Hyb input (Ring); Is the input for the audio that incomes 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. The Hybrid level knob works in the same way that with the lines connected directly to the D612's built-in hybrid.

2.7 Grounding against lighting

It is very important to follow these recommendations to avoid irreparable damage to equipment and people.

Download the hi-res diagram:
http://solidynepro.com/descargas/productos/Grounding_recomendations.jpg

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3.1   Overview

Solidyne D612 is an compact audio mixer designed for small radios. You can see different areas in the console:

- Input channels.
- Hybrid controls.
- Monitoring and talk-back.
- CUE
- MIC EQ
- Audio compressors and USB
- VU-meters

The bus **PGM** is the main output, that generates the mix that sent to the transmitter (or streaming).

The bus **REC** is a secondary mixing. Usually for recordings.

The operator can listen any signal before send it on-air; by pressing the **CUE** buttons.

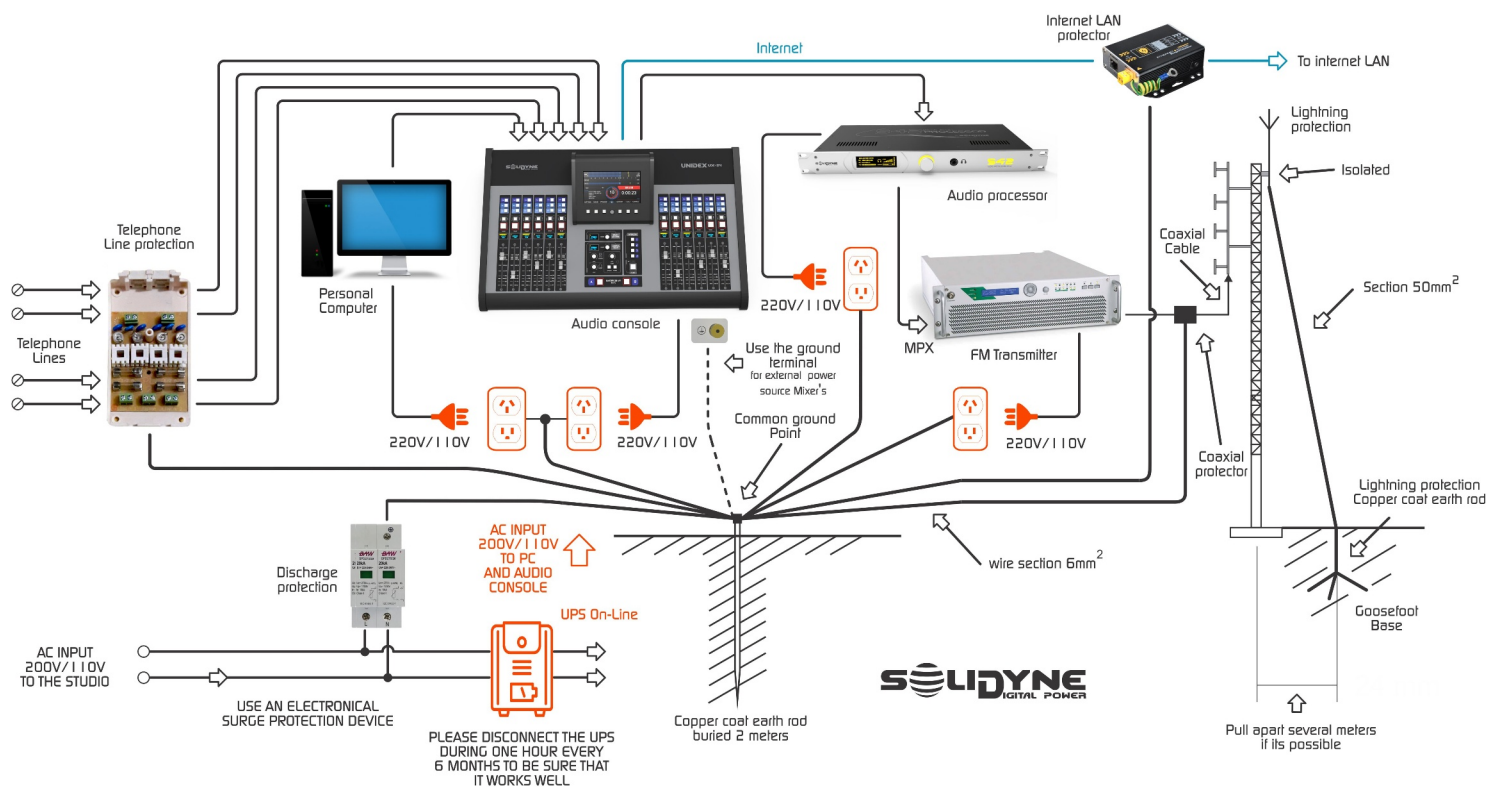Line channels will be on-air just opening the fader. MIC channels are turned on from Master MIC button. The switch "PGM/REC" assigns the output for each channel to the on-the-air mixing (PGM) or/and the recording output (REC).

Line channels have two inputs; which are selected from a switch located at the top of the panel.

- Channels **MIC 1** and **MIC 2** manages only microphone signal. The phantom 48V switch enables the phantom power to all the mics.

- The **microphone channels** can be routed to the **processing stage** (4 bands EQ and compressor) with independence.

- The **analog line channels manage** balanced (BAL) and unbalanced (LINE) stereo inputs.

- The **digital channels** manage the **USB** digital inputs (DIG) and one analogical stereo inputs (LINE).

- Additionally; USB offers direct recording to Hard Disk from PGM or REC.

The D612 LED's meters shows the recording and program level (average peak level) and the action of audio compressor.

3.1.1   Quick test

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:

- Choose a signal source, like a microphone, a CD player, etc. According to the 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).

- Press CUE button in the channel in which the signal is.

- Open the CUE fader in the MONITOR SECTION. The audio will have listened in the built-in loudspeaker.

To send on-air:

- Send the channel to PGM, so that the signal is sent to the main output.

- Open the main fader until reach 0VU in program VU-meter. The signal will be aired.

- 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.

Hybrid channel

The hybrid section manages **1 standard telephone line**, and **1 Bluetooth cell phone** (see "2.6.2 Connecting a cell phone") Both can be in simultaneous conference *(see 3.2.3 – Conferences)*.

Its use is simple and error free due to its logic of security and automatic control. The following instructions explains different scenarios of use.

3.2.1 Operation

When a call incomes, a blue LED blinks with the 'ring' cadence. The calling can be **answered** in two ways:

a) Using the telephone associated to that line

b) From the console, by pressing the HOLD/AIR button. When doing it, the line is taken and the LED stops blinking.

To answer the call from the telephone (connected to the RJ11 on the rear panel) pick up the phone and proceed as in normal communication. To put the call on the air, press the HOLD/AIR button and then hang up the phone.

To answer the call directly from the console, press HOLD/AIR. The console picks up the line and the ring's pilot light turns off.

In this condition, the caller receives the program signal, and will be on-air if the 'AIR LEVEL' knob opens. Therefore it is important to check that the 'AIR LEVEL' knob be closed when take the line.

To talk privately with the caller, press CUE. The caller is lietened on CUE speaker. In this condition (cue pressed) the caller not hear the program signal. To talk, press TALKBACK and use the talck-back MIC.

To put the call on the air:

- Release the button CUE.
- Turn the control 'AIR LEVEL' until reach the apropiate level.

The appropiate level for the communication is adjusted by observing the level meter "PGM", but also listening to the processed audio on the air.

To put the call off-air and resume in private, turn down the knob "AIR LEVEL" and press CUE to listen and Talkback to talk.

To **transfer the call to the phone set**, close "AIR LEVEL" to zero, pick up the phone and then release the "HOLD/AIR" button on the console. If the telephone was hung up when release the HOLD/AIR, the communication will lost.

To **hang up the line**, turn AIR LEVEL to zero and release "HOLD/AIR".

3.2.2 Using a cell phone

The cell phone connects to the Solidyne D612 via Bluetooth. The first time that a cell phone is used with the console, it must be paired as explained at "2.6.2 - Connecting a cell phone".

Once paired, the procedure used for the connection every time is simple:

- Make sure that the control "AIR LEVEL" is closed, and the button CUE pressed.

- Turn ON Bluetooth at the console. For this, press and hold "CELLULAR PHONE BLUE-

TOOTH" by 2 seconds (approximate) and release when the green LED (located in the VU meter panel) flashes. The LED remains flashing slowly indicating that Bluetooth is enabled.

- Turn on Bluetooth at the cell phone. If the cell phone was already paired, in few seconds the connection reestablishes and the cell phone stays linked with the console. Under this condition, incoming calls are transferred directly to console.

If the console is used exclusively with cellphone, ie, there is little or no use of the land line, you should leave the cellphone connected to the console and handle the calls from the hybrid.

When there is a continuous use of both lines, either alternately or simultaneously (conference) raises some peculiarities in the operation, detailed in "3.2.3 Conferences".

3.2.2.1 Incoming call

Is explained bellow 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' is listened simultaneously on the telephone and on the CUE speaker of the console (CUE button pressed)

- Check for "AIR LEVEL". It must be zero.

- To answer; make a fast touch on the button **"CELLULAR PHONE BLUETOOTH"** of the console (the calling can be answerer too from the cellphone).

After answering the call, talk to the caller, always using the hybrid of the console as follows:

- With the button **CUE pressed** the caller is listened on CUE speaker. Under this condition, PGM send to the cell phone is muted.

- Press and hold TALKBACK to talk with the caller.

- When CUE is released, the caller will listen to the PGM signal, but remains off-air due to the control "AIR LEVEL" stays closed.

- To put the calling on-the-air, turn up "AIR LEVEL" until reach the appropriate level on VU meters.

3.2.2.2 End the call

Pressing with short touch the "CELLULAR PHONE BLUETOOTH" button on the console, communica-

tion ends. It is also possible to end the call from the cell phone screen.

3.2.2.3 Resume the call on the cell phone

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

To transfer the call back to the console, return to the "Headset" mode. This operation differs depending on brand and model of cell phone.

Below shows a screenshot of a cell phone with Android, in the case of Bluetooth enabled.

The example shows the "Headset" icon, which transfers the call to Bluetooth device when it is powered on; and passes the phone when it is off. Other phone models can display two buttons; a button with dropdown options; or other combinations. Refer to the cell phone user manual.

3.2.2.4 Make a call

To make the call using the cell phone:

- In the cell phone, change the "Bluetooth" mode to "Headset" mode as explained in "3.2.2.3 - Resume 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 call on cell phone").

- Release the CUE button on the console. The cell phone receives the program signal, but remains off the air if "AIR LEVEL" is zero.

- To put the calling on the air, turn "AIR LEVEL" to the correct volume.

- To end the call from the console, make a short tap the "CELLULAR PHONE BLUETOOTH" button. The call can also be terminated from the mobile phone.

NOTE
The calling can be do from cell phone without leave the Bluetooth mode, using the console's talkback circuit. • The ring tone and the voice of the caller are listened on CUE speaker (button CUE pressed). • To dialogue, use the button Talkback of D-816. • To hold the call, release the button CUE. • To put the calling on-air, release "AIR LEVEL" • To end the call, make a short tap on "CELLULAR PHONE BLUETOOTH" button of the hybrid; or can hang-up from the cell phone screen.

Bluetooth advantages

Besides the convenience of being able to connect any cell phone without cables or special connectors, provides the advantage of placing up to 8 meters away, for example in a window to get the best reception.

But also enables new operational advantages for journalism, it is possible to connect the console in seconds with the private cell journalist for stories to people whose phone number is confidential and the reporter did not want to reveal (very common circumstance as journalists should be reserved with your contacts).

3.2.3 Conferences

When regularly using cell phone simultaneously with the landline, the user must be careful to OFF 'BLUETOOTH' ON CELL PHONE whenever a call ends. This way, if a new call comes while the landline is on the air, the call rings at the telephone. If Bluetooth were active, the call would sound on air.

The procedure for setting a conference depends on which of the two lines (landline or cell) is on the air. See the two cases below:

3.2.3.1 Being the cell phone on-air

Suppose the announcer is talking with a caller via cell phone; and the operator requires adding to someone else to the conversation:

- Make (or receive) the call using the telephone associated with the landline.

- Talk with the caller. Warn him that the call will go directly to the air. Remember that this is because the hybrid is being used by a cell phone, so operator can not put the second line on hold or use the talk-back circuit.

- Press "HOLD/AIR" to put the calling on-air, then hang up the telephone. The call goes immediately on-air (since "AIR LEVEL" is open).

3.2.3.2 Being the landline on-air

Suppose the announcer is talking with a caller via landline; and the operator requires adding to someone else to the conversation:

- When is not used, **Bluetooth must be off at the cell phone**, and so unlinked from the console. This way an incoming call (or is made) arrives to the cell phone with independence from the console.

- To transfer the call to the console, **turn on the Bluetooth on the cell phone** (see the cell phone user manual). Depending on brand and model of cell phone, the connection with the console can be immediately re-established; or it may be necessary to select from a menu 'Bluetooth devices' one to re-connect (in this case Solidyne BTM641) Once activated Bluetooth at cell phone, the call is put on-air immediately, staying in conference with the landline that is on-air at this time.

- To end the cell phone call, hang up from the cell phone.

- Then, turn off Bluetooth at cell phone. This is only necessary when there are a permanent use of the landline, to avoid that an incoming call to the cell phone accidentally appears on-air when the landline is on-air.

- Alternatively, to r**esume the call on cell phone**; using the cell phone menu it will switch to "Headset" mode, as explained in "3.2.2.3 Resuming the call on the phone"

3.2.4 Send PGM signal to the phone

The internal built-in hybrids dont'n have a level control for the signal sent to phone line (aka "return" from the point of view of the remote caller). This is because the signal sent to phone line is processed by AGC, peak limiter and filters. There-fore the send level is automatically adjusted during the transmission and its level is the maximum allowed by the modern digital telephone exchanges.

To check the send level to phone line, use an oscilloscope in parallel with the phone line and verify that the signal level is 2V peak to peak. Levels over this value can cause like intermodulation distortion in the PGM signal.

For this reason the Solidyne hybrids use a processed return channel, to avoid distortion at the on-air signal. The send signal is limited to 400 – 2.200 Hz in order not to distort the signal on 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 the band is narrow. This should not worry because it is a subjective sensation that does not affect the intelligibility of speech.

3.3 Monitor section

3.3.1 Monitors and headphones

In the section MONITOR you can see the following areas:

- *Studio Monitor: here you define what signal you will listen at the Studio monitors and headphones.*
- *Control Room Monitor: define what signal you will listen at the Control Room monitors and headphones.*
- *CUE: set the level for the CUE signal.*

✍ The built-in speaker can be disabled and the **signal CUE** can adds to the Control Room Headphones. It requires to change an internal connection (see 2.4.3 Motoring outputs).

There is a knob to manages the level of headphones and monitor outputs for the Studio. Usually the monitoring level adjusts from the console for a comfortable listening in the headphones. If needs individual level control for each headphone; a headphones mixer (Solidyne Studio-box or similar) can be used.

The **Studio speaker monitors** have their own gain control, since the console's output was designed to operate with powered speakers. When the microphones are on-air studio monitors are muted, to avoid feedbacks loops.

Like the Studio section, the Control Room has a unique knob for the headphone and the speakers. The Control Room monitors must be powered units (active speakers).

Routing switches allows to choose the signal source for monitors and headphones. The assignments for Studio and Control Room are independent. The options are:

- **PGM** *to listen the console* output.

- **REC** *to listen the channels that are assigned to recording output.*

- **AIR** *switches to an input designed to connect an external tuner in order to* **monitoring the transmission from the air**. *This is the right way for monitoring; recommended for all situations.*

3.3.2 Use into integrated Studios

There are many radio stations in which the mixer console and the host, locutors and journalists are located in the same room. It avoids the double room separated by a window; reducing the building costs and the acoustic treatment.

In this cases, connect only one active speakers to the output of Studio Monitor. This way, when the microphones are on-the-air, the speakers are muted. Connect the headphones for talent and guests to the output Studio Headphones. Connect the operator headphones to the output Control Monitor (see at next).

3.3.3 Previous listening (CUE)

The D612 has an internal loudspeaker for CUEing. Each channel has a button "CUE" that allows to listen the signal present in that channel with the channel fader closed (PFL).

When the mixer console and the microphones are placed into a same room (hot studio) the built-in speaker must be disabled and the CUE signal sent to the operator (Control) Headphones. To make this, an internal connection must be changed, as was detailed in *2.4.3 Monitoring outputs*.

If CUE is pressed in several channels, the signals are mixed. Rotary knob CUE manage the level of the CUE signal.

3.3.4 Talk back

The talk-back circuit allows the operator to talk with the speaker while he is into the studio.

In order to talk, press the Talk-back button (see monitor panel drawing) **.** The Control Room monitors will be muted, to avoid feedbacks loops. Your voice will be listened inside the Studio, by the left channel. The right channel stays with the on-air audio, so that speakers and journalists do not lose the On-Air reference. At the same time talk-back microphone will be listen in all studio headphones.

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

> ✎ The level of the talk-back microphone comes calibrated from factory, and usually it is not necessary to modify it. If you need to change it, there is a preset called "LEVEL" to make the adjustment.

Audition level for Studio loudspeakers and headphones

Audition level for Control Room loudspeakers and headphones

CUE level. Set the level for the CUE signal.

PZM MIC. For **Talk-Back** and hybrid

Assign **PGM** or **REC** to the Studio monitors

Change the monitoring between **external input 'ON AIR'** and PGM/REC (Requires an external tuner)

Talkback volume

Talkback to Studio

3.4   Microphone channels

Gain →

Enable/disable the phantom power (48V) for MIC-1a and MIC-2a.

The processed signal can be sent to PGM and REC. To the left, the switch cancels the sending.

Previous listening (Cue)

The gain must be adjusted in order to the main fader works over -30 dB indication. **OVL** indicates overload, thjat is to say, excesive input gain.

4 bands EQ. This stage has a dynamic range compressor compresor that begins to work over 0 VU. The indicator "MIC compressor" shows its action.

The channels can be assigned direct to PGM or be send to the processor stage (EQ)

Main fader. To enable the microphones press the **MASTER MIC button,** located at the right on the console.

The apropiate operation level is mantaining the peaks at 0 VU

3.4.1   Features

There are two microphone channels. The **"Phantom 48V"** is common for both channels; but only MIC-1a and MIC-2a gives phantom voltage. Remember that all dynamic microphones can connect to a line with phantom voltage, without risk of damage; because internally they have a transformer that disconnects the DC. So it is possible to **mix dynamic and condenser** microphones.

All the channels have a **gain knob**. It must adjust so that the main fader works over the overload **"OVL"** indication (- 30 dB). Remember that in all audio consoles, excessive input gain reduces the dynamic range**.** If you work with the main fader below this level, the signal from the previous stage must be very high and the peaks will be clipped.

3.4.2   Processing

Each MIC channel can be sent directly to the air (PGM) or the processing stage (EQ + compressor). As well, the processed signal can be sent to the air or to recording out.

Equalizer

The equalization settings are the same for all assigned channels. The figure shows the equalization curves. These curves are designed to improve the quality of the human voice. The ends of the spectrum work in shelving way, with 15 dB action.

As reference we said that the LOW (or bass) control offers "body" and "weight" to the voices; whereas the HIGH one 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 plopping effect are concentrated, or there is excessive resonance in certain masculine voices. The mid-high band is located at 5 KHz, because around this frequency usually appear the problems with the "sss" (excessive emphasis the sibilants sounds, like "shhh")

Compression

The compressor maintains constant the peak level of the human voice. It starts to work when the signal surpasses the 0 VU level. Below that level the dynamic range is not affected. D612 has a very steep compression slope and fast attack and recovery times.

Indicator "MIC COMPRESSOR" shows in dB the **reduction** applied to the signal.
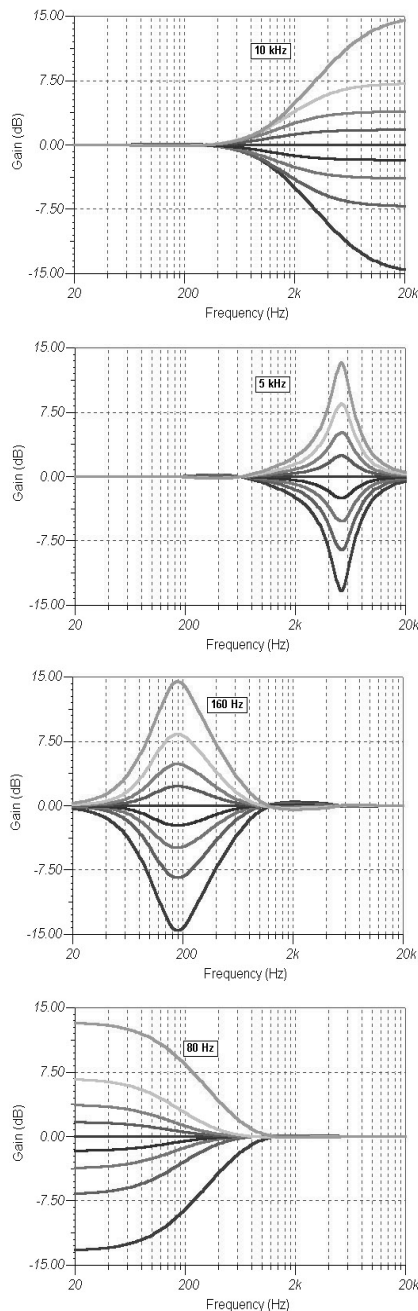
3.5　Line channels

Solidyne D612 mixer console has 3 stereo LINE channels. LINE channels have two inputs that are selected from the front panel.

- **Channels 3 and 4** have balanced stereo inputs (DIN-5 connector) and unbalanced stereo inputs (RCA).

- **Channel 5** are digital, with direct connection to the PC via USB. It is equivalent to have an external sound card. Additionally it gives one USB send to the computer (PGM or REC). The secondary input is analog unbalanced stereo (RCA).
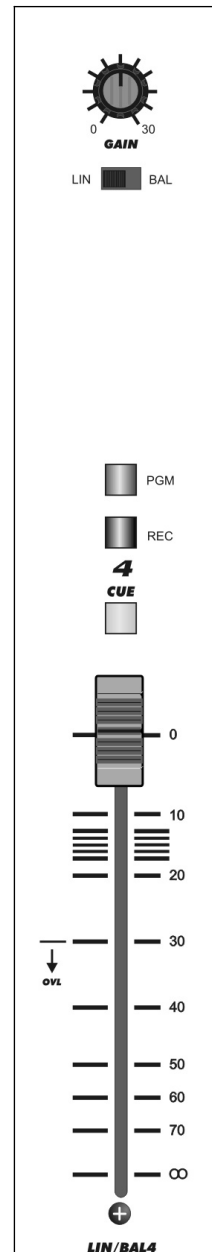
LINE channels have gain knob. For the case of USB channels, there is not input gain. Level adjusts from Windows audio mixer.

Set the input gains in order to the main faders, with an nominal level at input, works always above the OVL zone, to avoid overload.

Connections are explained at *"Chapter 2 – Installation and wirings"*

To put a channel on-the-air; assign the channel to PGM and open the fader. The correct level is when the peaks at 0VU continuously.

Main faders works with **ETM-VCA technology** (Electrometric-Voltage Controlled Amplifier). The fader only manages DC, that control amplifier with variable gain.

This method guarantees very low distortion and is noisly-free. For more information about the ceramic fader technology, please visit:

www.SolidynePro.com.

3.6 Recordings

D612 consoles have two stereo output channels. The main channel is denominated Program (PGM) and used for the On-Air signal.

For recordings, REC bus is used. The user can make recordings while, simultaneously, the console is on the air. We will see like example the procedure to make a recording in the computer whereas a musical program is on air.

NOTE: the two telephone hybrids of this console are units "aired" and therefore can not be assigned to the channel REC. If you need to record an interview you should connect to the console a hybrid external; Solidyne type HL-202.

3.6.1 Direct to Hard Disc recording

Remember that the main output (PGM) and the recording output (REC) are available in digital format, through connections USB.

Once connected, the outputs appear in Windows© like: (Control Panel/Audio and Sound Devices/Audio)

"USB Recording device": corresponds to channel USB 5 and gives the PGM or REC signal.

***On the PC:* configure your recording software** to use an USB recording device.

At the console: route the channels that you want to record to REC bus. The others remain assigned to PGM. You must proceed as follows:

a) In Monitor Section assign the Control Room monitors to the recording bus (REC). If you want you can send REC signal to the studio monitors too.

b) Assign to REC the channels required for the recording (for example MIC-1 and Line-4). When a channel is routed to REC it is removed from Air

c) Meanwhile the radio still on air through, for example, the channel 5 with a musical program.

✍ | Channel-5 can be used for playing with independence of the assignment to USB OUT (PGM - REC).

Now makes the recording on the PC. In order to activate the microphone, open the fader of MIC-1 channel.

The recording level is verified in the REC VUmeter of the recording software, and it adjusts from the console. You cannot change the input level from software (Windows Mixer).

> **USB RECORDING DEVICES DO NOT HAVE GAIN CONTROL FROM THE SOFTWARE**

The speakers will listen to the recording USING HEADPHONES or in loudspeakers. When the recording ends, it can be listened using the same REC bus, assigning to REC the channel in which the PC is. At this point; BE CAREFUL OF NOT PRODUCING A FEEDBACK LOOP IN THE OWN 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, reset all the controls to the original position, to return the console to the normal way of operation.

3.6.2 Recording the phone lines

Hybrids of the D612 signal sent only to program, therefore, only be recorded telephone communications PGM signal recording.

To record the PGM output, you must use a wire tap "Y" for a "copy" of the program output, simply connecting two parallel cables.
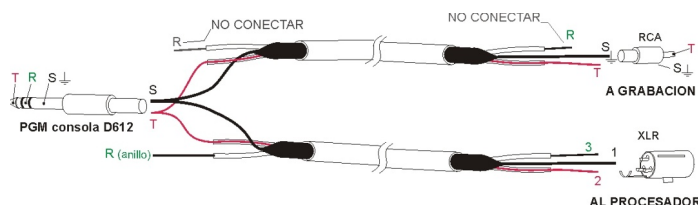
"Y" cable - Plug 1/4" BAL to XLR BAL and unbal RCA, to one input channel. For the other channel you need a second "Y" cable.

4.1 Introduction

The models D612/IP includes an RJ45 port that sends or receive a streaming. This feature allows:

- Link the console with the transmitter. With high-quality MP3 or PCM using a point to point wireless link.

- Send or decode an RTP streaming from others devices.

By default, the firmware is the **STL** (Studio to Transmitter Plant link) in mode ENCODER. The user can upgrade or change the firmware to model **Upstreamer,** in order to connect the console to Icecast/Shoutcast servers.

4.1.1 Firmware upgrade

The firmware can be changed for:

- Update to a new version of STL

- Change the firmware STL to Upstreamer version, to stream to shoutcast/icecast servers.

The procedure to change the firmware, and the settings for the Upstreamer mode are detailed in the following paper:

Solidyne equipments with the optional AoIP module – Firmware upgrade tutorial and shoutcast settings.pdf

4.2 Power source

The IP module receives power from the USB connection, so it is only available if 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 can be connected to D612.

4.3 STL link

In studies, the console works as ENCODER D612/IP to establish a unidirectional link (half-duplex) between Studios and Transmitter Plant, generating and transmitting an audio streaming.

At the transmitter plant, a device streaming decoder receives the transmission (eg

Solidyne ADA102) and sends the audio processor. Another option is to have a decoder processor with streaming audio, or a computer running software that play streaming (eg. VLC Player).

4.3.1 IP access

The configuration options D612/IP Solidyne are accessed using a web browser. The unit is factory set in "Dynamic IP", so that when connected to a LAN, gets an IP address via DHCP (the router assigns an IP). The procedure is as follows:

Step 1

Connect the console to the power supply. Connect a computer via USB to the D612 to enable the IP module.

Step 2

Connect the console to the network via a standard cable. The network must have a router, so that the console assigned to an IP address via DHCP. D612/IP You can also connect directly to a modem-router, as it usually also assign an IP via DHCP.

Step 3

The TCP module will announce the IP assigned. To listen to the IP address, proceed:

- Turn "BAL-LIN" Channel-4 to "LIN-2".
- Press the CUE button in the Channel-4 and lift the CUE speaker level.
- Disconnect and reconnect the USB cable from the console (computer must be on).

Take note of the address announced, it will be needed later. This operation is only needed during installation.

Step 3

Once you get the IP, is ready to start working. The green LED on the rear panel (RJ45) is blinking.

Can not find a DHCP server then the D612 scans the network for a free IP address (this can take a few minutes).

Step 4

Open a web browser (eg Firefox, Internet Explorer) and enter the IP address announced. The Control Panel D612 is displayed.

4.3.2 Destiny IP address

The status screen indicates the IP port configuration. The module is factory-configured as Studio Encoder (option "Location" → 'Studio Encoder').

Step 5

To verify the configuration, access the "Configuration" and in the left menu choose **"Basic settings"**.

BASIC SETTINGS

OUTGOING STREAM

Stream Method	URL	Port
Push(RTP)	192.168.0.30	4050

- **"Stream method"** should 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 your 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 DECODER (static IP assigned by your ISP). When the data packets reach the router / firewall on the other end, should be re-directed to the DECODER IP (eg 192.168.0.30).

As the D612/IP transmitted to a specific IP address and port, in transmission plant all packets arriving at that port on the router address should be forwarded to the DECODER, or a particular computer, which will turn into audio. Identify which packets should be sent using port forwarding.
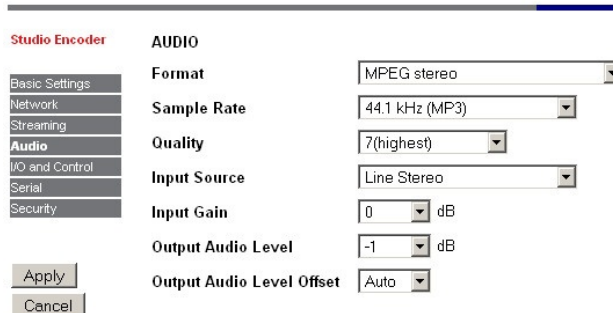
4.3.3 Audio settings

Step 6

Go to Config → Audio

Input source: Default is *"Line Stereo". Do not change it*.

Format: Streaming format. Default values are:

- Format: **MPEG stereo**
- Sample rate: **44.1 KHz** (**MP3**)
- Quality: **7 (highest)**

Studio Transmitter Link

These settings produce a stream of 192 kbps. Should only be changed if the bandwidth of the Internet connection is limited, or if a link is available that enables point-to-point transmit uncompressed (PCM).

4.3.4 Using a microwave digital link

A console connected studies D612/IP broadband Internet to cover any distance from the studios and the transmitter.

In short distances the option is point to point RF link transmiting uncompressed audio (PCM 16 bit/48 KHz).

It uses a microwave link for 5.8 GHz (or 2.4 GHz in some countries) using the standard 802.11.x. This band is available in all countries and does not require special authorization. It can cover up to 45 miles if there are no obstacles between the extremes. Logically encoded audio can be transported and for special applications supports bidirectional connection.

For more details, please contact us at our email (info@solidynepro.com).

4.3.5 Decoding using a computer

To receive the transmission using a computer, you should use playback software that supports streaming and RTP protocol. Solidyne recommend using VLC Player (http://www.videolan.org/vlc/).

For the transmission, set the DX816/AoIP to work with **PCM16 @ 44.1KHz;** or any MP3 mode (see 2.7.1.3 – Audio settings).

To start to play in VLC go to *Medio –> Open net-work*

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

Example: RTP://0.0.0.0:3030

In where RTP is the protocol used by the Solidyne D612; 0.0.0.0 means *from any incoming IP* and '3030' is the port used*. Remember to enable your computer firewall to allow incoming streaming trought port 3030 (or the port that you use).*
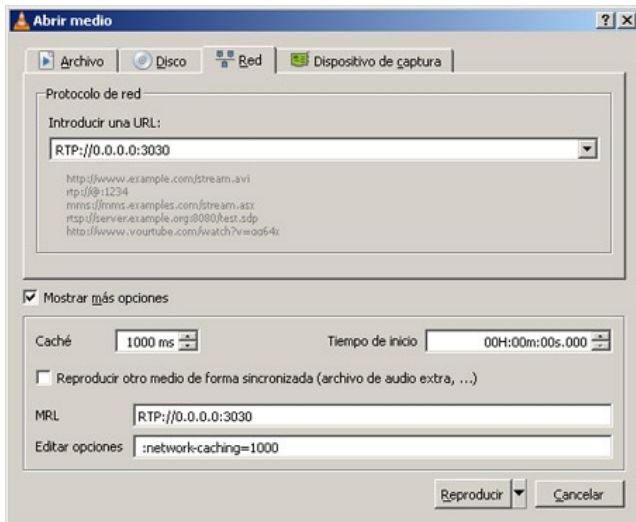
NOTE: *Some versions of VLC Player also supports the string* RTP://@:3030

4.4 Decode an incoming streaming

The D612/IP firmware can be settled to decode an incoming streaming; for, example, generated from other Solidyne mixer (series 2600/IP; DX816; DX822 and ADA102 coder).

At factory, D612/IP is configured to work as coder, generating the streaming, To receive an remote streaming, D612/IP work mode must be changed to DECODER.

To set this mode, proceed:

1. Using any WEB browser, access to the D612's IP module (please refers to 4.3.1 – IP access).

2. At the top menu, click the option **"Location"**.

3. By default, D612 works as CODER, so it generates an 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 ADDERSS.

4. Enter to the option **"Settings → Streaming"**.

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 type of audio streaming.

Buffer size is expresed in miliseconds. For uncmpressed 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 (see "2.3.5 - About audio options").

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

4.4.1 Decoded audio from the incoming streaming

The audio signal decoded from the incoming streaming **is mixed with the auxiliary input of Channel-4** (LINE-2). To send on-air proceed as follow:

1. Switch **Channel-4** to **LINE-2** (unbalanced input).

2. Leave the aux input (RCA) unconnected.

   NOTE: To use the audio signal from the incoming streaming, leave unconnected the unbalanced inputs (RCA) of Channel-4, since both signals are mixed internally.

3. Raise up the fader of ´Channel-4' to send the incoming streaming on-the-aire.

4.5 Others connections

4.5.1 Full-duplex link with unknown IP at the Studios

This case is a full-duplex link, but only the transmitter IP is known. The coder at the Studios starts the communication to that IP address.

At the console A:

In the main page, in option "Location", set the device as "Studio Encoder/Decoder". Go to "Configuration → Basic settings" and enter the IP address or URL for send and receive.

Set the outgoing stream method as "Push(RTP)". In URL enter the IP of decoder (if consoles are in different locations, th eIP must be the external IP ad-

dress that router redirects to Console B). The Port number is the the "Local Port" value assigned to the Console B decoder.

Set the incoming stream as "Pull(BRTP)" since the IP address of Console A is unknown. In the field URL enter the IP of the Console B (decoder) and the same port number assigned to the decoder.

OUTGOING STREAM

Stream Method	URL	Port
Push(RTP)	192.168.0.202	3031

INCOMING STREAM

Stream Method	URL	Port
Pull(BRTP)	192.168.0.202	3030

In a full-duplex link, <u>both units must work with the same PCM audio settings</u>.

At the console B

The Console B receives the streaming from the console A. Set "Outgoing stream" as "Pull(BRTP)" since the Console A started the connection. IP address is zero and the Port number mus be the used in the section "Incoming stream" at Console A.

Set "Incoming stream" as "Push(RTP)" with IP address zero, since the Console A starts the connection to the B (decoder).

OUTGOING STREAM

Stream Method	URL	Port
Pull(BRTP)	0.0.0.0	3031

INCOMING STREAM

Stream Method	URL	Port
Push(RTP)	0.0.0.0	3030

4.5.2 Link with repeaters

In this mode the coder send streaming to several devices simultaneously. This configuration is common when you have a static IP to the *"multicast"* group and requires a half-duplex link. In both ends the AoIP modules are set as "Multicast" *(Configuration → Basic settings → Stream Method)*.

Multicast is a protocol designed for maximum efficiency of the networks when required simultaneous transmission from one point to multiple destinations. Both the transmitter and receiver use the same IP address, which should be in a special range of IP addresses for a multicast. This range includes addresses from 224.0.0.0 to 239.255.255.255.

To implement this scheme, see your network administrator, because you need to configure the multicast routers.

At the coder

In this case the coder transmit to a multicast IP address. Coder works in mode "Multicast" *(Configuration → Stream Method)*.

At the decoder console

The decoder aslo works as "Multicast". In the field URL you must define the same multicast IP used in the coder.

5.1   Overview

Solidyne HD3 and HD5 are an optional accessories that groups in a single box all the signals for monitoring needed into the Studio:

- **5 headphone's** in HD5 with independent knobs (3 headphones in HD3).
  - Double bus for monitoring from two sources.
  - Output for **active speakers** with a gain control.
  - **Tally light**
  - **Timer/Clock** (only in HD5)
  - **Talk-back**, to talk with the Control Room.

5.2   Connections

Usually the HD3/HD5 dock is placed on the table; but a good choice is fix it to the wall when the table is next to the Control-Studio window.

To active speakers   Secondary audio input   To DX816 Studio Monitor Out
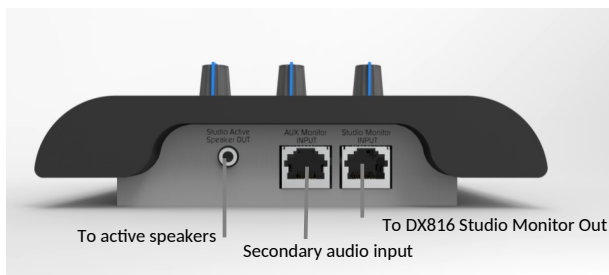
Figure 1: HD3 rear panel

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" (mini-jack).

5.2.1   Connection to the mixer console

The HD3 and HD5 accessories were originally designed for the advanced console series. But now it is possible to use them with the D612 by means of a special adapter cable provided by Solidyne. It is connected in the following way:

1.- Tally light: Is a tubular connector. This connector plugs to Tally output of the mixer to take the on-air signal and activate the timer and on-air light of the StudioBox. The second connector (pass-thru) is for connect the standard tally lights.

2.- Studio speakers: Takes the monitoring mix for the studio speakers. At the Studio, the speakers (active speakers) will connect to the StudioBox.

3.- Studio headphones: Takes the mix for the studio headphones. Up to 5 headphones can be used connected to the Studio Box.

4.- Reverse talk-back: Is a 1/8" Jack TRS that sends the signal of the HD3/HD5 built-in microphone. This connector allows to plug a small active speakers located at the Control Room, to allows the host can talk to the operator in the Control Room. The operator also can use the CUE of microphones channels to listen to the Studio voices.

5.- Power source: The HD3/HD5 feeds from an external power source 110/220V 1.25A.

Figure 2: The DB9 connector was replaced by RJ45 in new models

5.2.2   Connections at the Studio

Headphone outputs use ¼" stereo TRS (jacks). Any combination of headphones can be used (16; 32 and 64 ohms).

Loudspeakers output use a 1/8" stereo TRS (minijack). Take in mind that this is a **line level** output, designed to use with active speakers.

5.3 Using the HD3/HD5

On the Studio Headphones

HD5 manages up to five headphones with independent level knobs. HD3 allows to connect up to 3 headphones, but phones 2 and 3 shares the volume knob.

The maximum volume on headphones depends on the *Studio Headphones* level at the console. The signal listening also depended on the selection done in console.

On the Studio Monitors

Loudspeakers output has a level control, but the maximum level and the signal source depended on the console selection.

If the loudspeakers are connected directly to the console's speaker outs; "Loudspeaker" doesn't take effect.

On the Studio Talkback

The white button has two functions:

1. When the **microphones are off-air**, press the button to speak to the Control Room. The HD3/HD5 has a built-in microphone for this purpose.

   At the Control Room, the operator will hear to the talents using **active speakers** connected to the HD3/HD5. To answer, the operator will use the standard talk-back circuit of the console.

2. Being on-the-air, the white button illuminates in red indicating that microphones are "on-air". Obviously, in this condition talk-back feature is disabled.

✎ | The preset located below the MIC allows to adjust the microphone gain.

On Timer / Clock (only for HD5)

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

The "counter mode" can be disabled; quitting an internal jumper. Open the back cover of the Solidyne HD5 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.

CHAPTER 6

In order to obtain great results with D612 consoles, we recommend to follow these steps.

6.1 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**.

D612 consoles use the new Conductive Ceramic fader technology with more than one million operation guarantee (usually 15 years of use). Please see more information at www.solidynePRO.com

This faders do not need maintenance nor cleaning.

6.2 Preventive Maintenance

The Solidyne D612 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 D612 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 headsets directly to console. When the operator's changes frequently its headsets, use an external patch panel to avoid wearing console jacks

6.3 Spare parts

Faders for all channels are connectorized. They are easy to replace, and which are mounted to the chassis with two screws and have a polarized plug to connect. In normal use, do not require faders changed throughout the life of the console, but if the environment was free of dust, or the operators were moving unnecessarily (syndrome "nervous finger"), will eventually require replacement   .

Solidyne provides all necessary parts such as buttons, knobs or faders. Write to: info@solidyne.com.ar

6.4   Service Manual

Service manuals, with the electrical circuits and components layout, are sent by email to Authorized Solidyne Partners 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**. To be visualized in the screen of a computer, the technician will require of a USB key and software to be able to decode and see the documentation. This software is sold separately and is used also for all the Solidyne products.

Solidyne brings full support to technicians during the repair work. Please contact us at:

- info@solidynePRO.com
- www.solidynePRO.com

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5.5 Tech specs

Audio inputs
4 balanced MIC inputs (XLR)
2 balanced stereo Line inputs (DIN 5)
3 unbalanced stereo Line inputs (RCA)
1 Digital stereo Line input (USB)

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

Phantom power

Two mic inputs have 48 V phantom power supply, switchable ON/OFF.

Analog outputs
2 bus stereo outputs; PGM (1/4" TRS plug); REC (RCS connector)
Both at +4dBm = 0VU
PGM balanced: Max level +26dBu (10K), +20dBm (600 ohms)
REC unbalanced: Max level +20 dBu
External Hybrid: unbalanced mix minus out + 4dBu

Digital Input/Output levels

USB digital In/Out; Input to channel 5; Output selected PGM or REC. It uses the K-15 recommendation for broadcating. 0 VU = - 15 dB Full Scale level

Monitor & Hybrid Outputs
1 Stereo Studio, 0 dBu for active speakers, muted
1 Stereo Control, 0dBu for active speakers, muted
1 Stereo Studio Headphones, with Distribution AMP up to 5 headphones, + 10 dBu
1 Stereo Control Headphones, + 10 dBu
1 External Hybrid send (MIX-Minus) +4 dBu / 10K

Cue monitor
Internal CUE Monitor Amplifier; 250 mW , with built-in speaker.

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

Frequency response
20-20.000 Hz +/- 0.25 dB (LIN to PGM)
20-20.000 Hz +/- 0.5 dB (MIC to PGM)

Mic equalizer
Four Bands MIC EQ with In/Out switch; 80 Hz top flat curve, 160 Hz bell shaped, 4 kHz bell shaped, 8 kHz top flat
Action + 15 dB / -15 dB

MIC compressor
Low distortion MIC compressor with In/Out switch. 20 dB max compression. Attack time < 10 ms. Threshold = 0 VU
Compression ratio: < 2 dB variation for 15 dB change MIC level

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

Dinamic range
From Line to PGM > 90 dBA (CD quality)

Crosstalk
PGM-REC > 65 dBA @ 1 kHz

Distortion
From LINE to PGM out < 0.05% THD @30-15.000 Hz.

Phase
From Line to PGM, < 3º L&R @50-10.000 Hz

Stereo tracking
Below 0.2 dB error L/R in fader range 0 to 40 dB

Telephonic hybrid
Two channels telephone active Hybrid: One phone line plus one cellular (Bluetooth linked)
Frequency Response: 200 - 5.000 Hz
Noise: > 60 dBA S/N
Rejection: > 35 dB rejection
Rejection adjust in front panel preset
Preference atenuator: 12 dB local announcer interrupt priority

Telephone input lightning discharge safety
Telephone Hybrid inputs is transformer floating to meet the Public Telephone service isolation standards. It is protected with SIOV Varistors against soft lightning discharges. External gas discharge protection is recommended

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

VU-meters
2 x LED vumeters for PGM. Cuasi-peak reading.
Microphone compressor indicator 0-15 dB

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

Power
Switching external power supply 90-240 V, 20VA [28V@0,5 A].
Tubular connector 1.8 mm (internal pin).

Dimensions & Weight
Wide: 405 mm; Deep: 385 mm; Height: 95 mm at rear turret
Net Weight: 4,7 Kg

AoIP - Optional TCP / IP Digital Streaming Output

Ethernet RJ45 connection to a standard LAN (Option /**AoIP**).

Used for sending D612 output to transmitter plant with MP3 Internet streaming or using a 5,8 GHz 802. Link Also can be used for **AoIP** connection with another console located inside or outside studios.

It can work in linear **PCM** at 44.1 & 48 kHz or **MP3** VBR.

On-Air information using GPIO for remote processors.

AoIP Delay and user modes
• MP3 Layer 1 (32, 44.1 and 48 kHz)
• MP3 Layer 2 (16, 22.05 and 24 kHz)
• 16bit PCM uncompressed (8,16,32,44.1 & 48 kHz)
Full duplex (transmitting and receiving)

Delay using PCM at 48 kHz Fs (recommended) Less than 5 mS Delay using MP3 streaming. Average 100 mS

Notes