Active direct injection unit for stage and studio

This low cost DI box offers performance and facilities only found on the most expensive units. It converts a high impedance unbalanced input to a low impedance balanced output.

by ROB EVANS

The so-called "DI" box has become an essential part of studio and sound reinforcement equipment, its humble features found hiding in many a dark stage corner.

Although the expression "direct injection" may sound like a technique used in Artificial Insemination, (as some have suggested!), it is simply the process of connecting an electric or electronic musical instrument directly to the balanced microphone input of a mixing desk. This system avoids the use of a microphone, which is often placed in front of a loudspeaker of the instrument's stage amplifier. The signal received by the mixing desk is of a much higher quality with a DI box, for it avoids the non-linearities of the loudspeaker to microphone signal path.

As a bonus, more than ten of the described DI's could be built for the price of a high quality microphone. Good for the pocket as well as the ears!

Designed for the rigours of professional use, the DI is housed in a diecast box and uses rugged flush mount switches.

A DI box is designed to electronically "look" like a microphone to the input of the mixing desk, yet essentially offer no loading or degrading effects to the instrument source. Hence the unit is designed to have a high input impedance, while providing an output level and impedance similar to a microphone.

Balanced lines

High quality mixing consoles and microphones use the balanced line principle for their interconnection. This system enables very long cable runs to be used, with little losses or interference problems in a "hostile" environment (e.g., a strong mains field from a lighting dimmer rack)

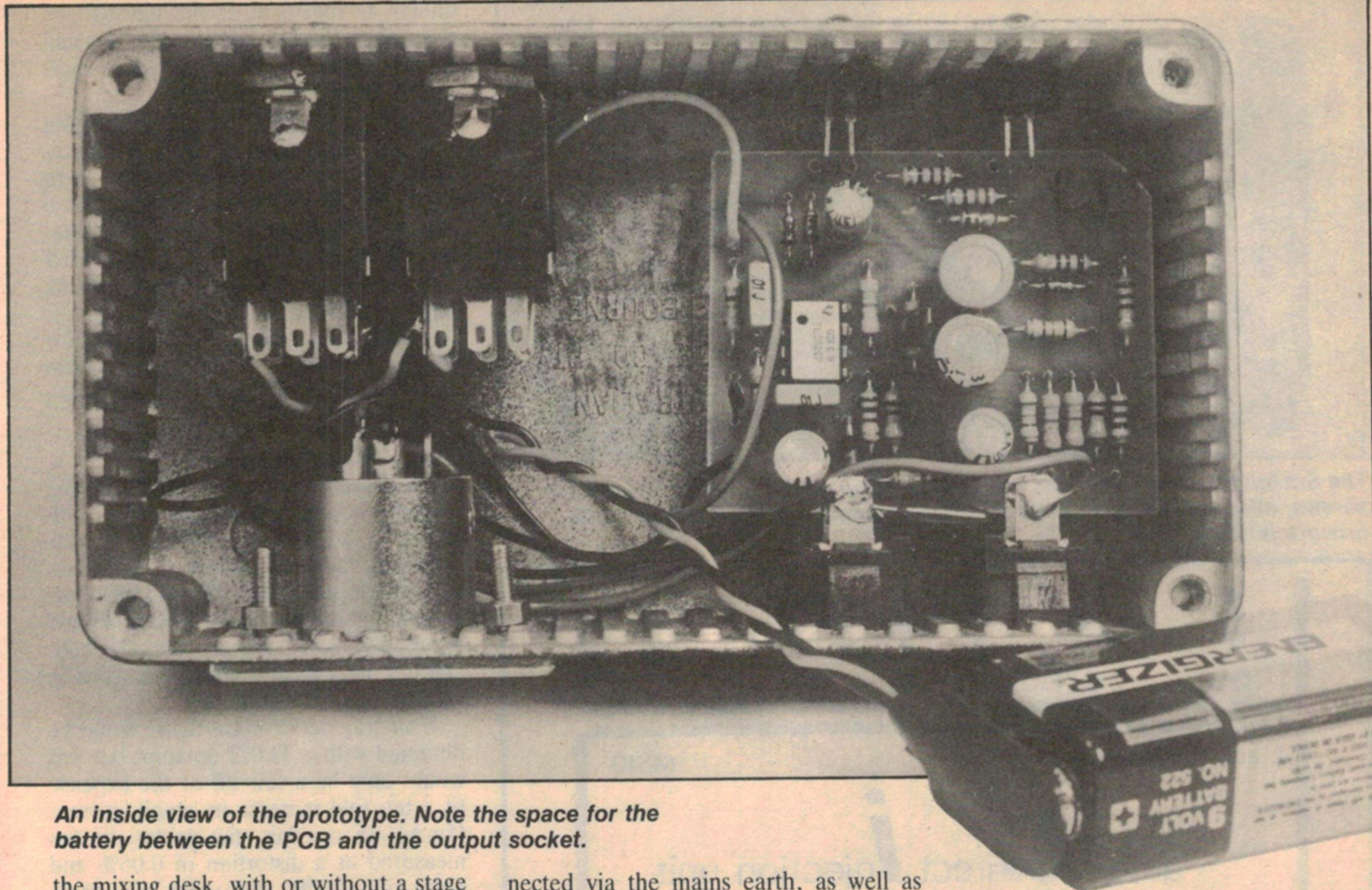
The standard balanced audio system uses a two core shielded cable terminated in XLR-style three pin connectors. Pin 1 is connected to the shield, while Pins 2 and 3 carry the signal, Pin 2 being 180 degrees out of phase with Pin 3.

This "antiphase" signal is received by a differential or subtracting amplifier, which will produce an output proportional to the *difference* in signal between pins 2 and 3. That is, any signal *common* to both lines will be ignored. Such is the case for induced hum, which will appear on both signal lines and thus be rejected by the differential amplifier.

The transmitting and receiving of these balanced signals has traditionally been achieved with transformers. This method is slowly yielding to the active balancing system, utilising low cost op-amps. Appropriate high quality transformers are very expensive, and can suffer from distortion and lack of shielding.

DI features

For a relatively simple device, the DI box requires a number of features for it to be completely effective in serious studio and stage use. It is generally used to couple a bass guitar or synthesiser to

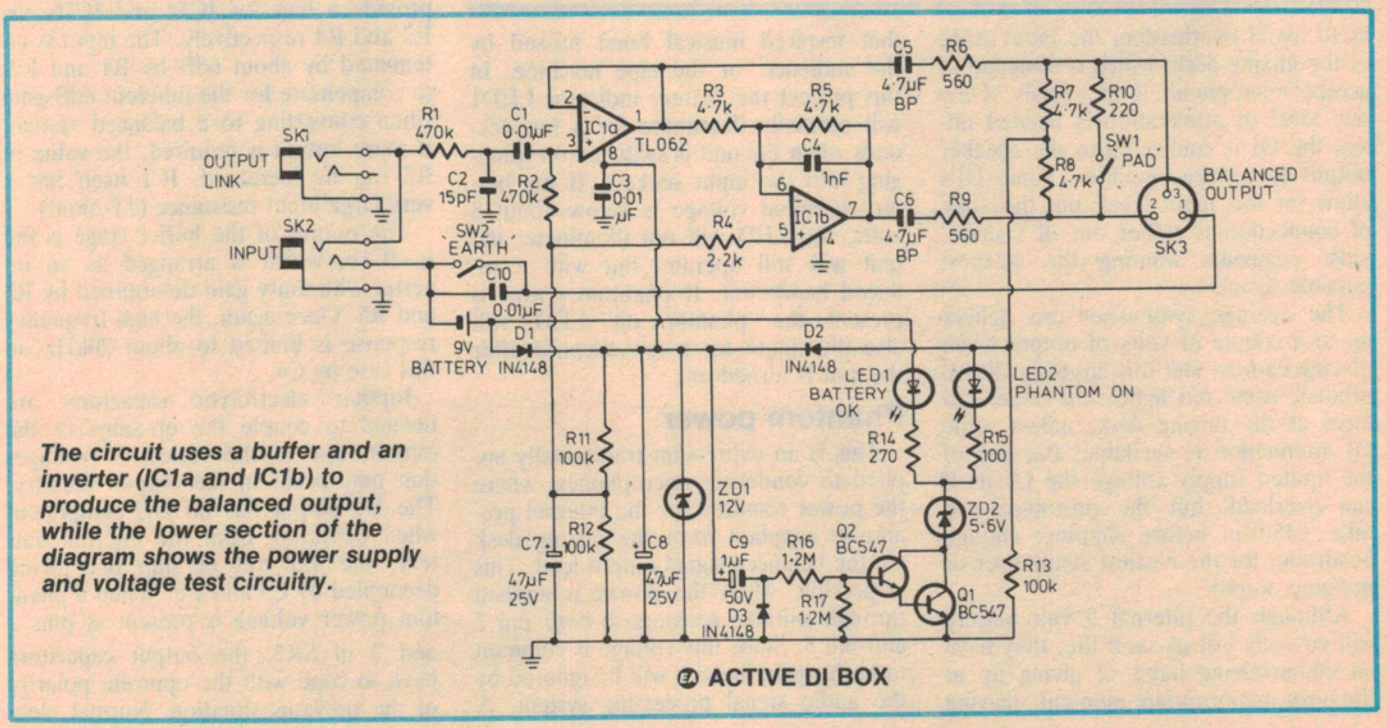


An inside view of the prototype. Note the space for the battery between the PCB and the output socket.

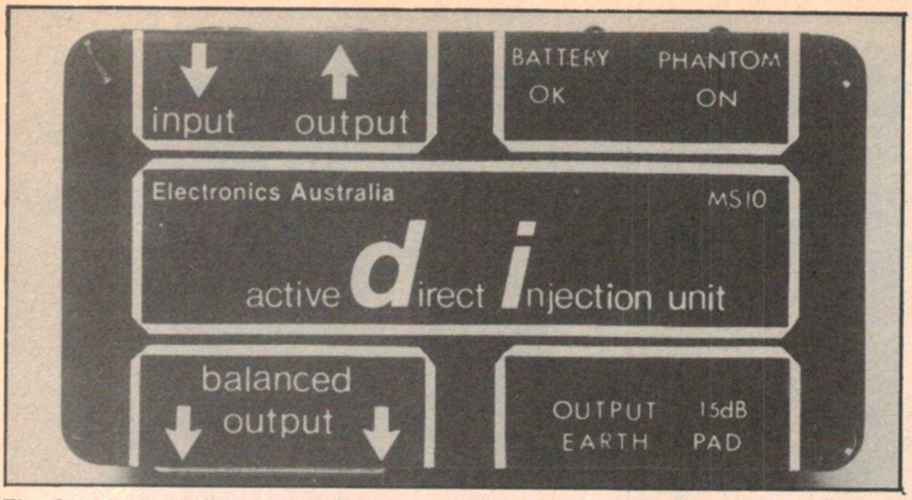
the mixing desk, with or without a stage monitoring amplifier.

If a stage amplifier is used, an "earth loop" situation can easily develop, often causing a loud hum in the system. The "loop" in this case is due to the desk and instrument amplifier being connected via the mains earth, as well as the signal cable shield (Pin 1 of the XLR connectors). A switch is included in the DI box to disconnect pin 1 from the input earth when this situation occurs. Naturally, when the instrument (e.g. bass or acoustic guitar bug) is connected only to the DI box, the earth must be continuous.

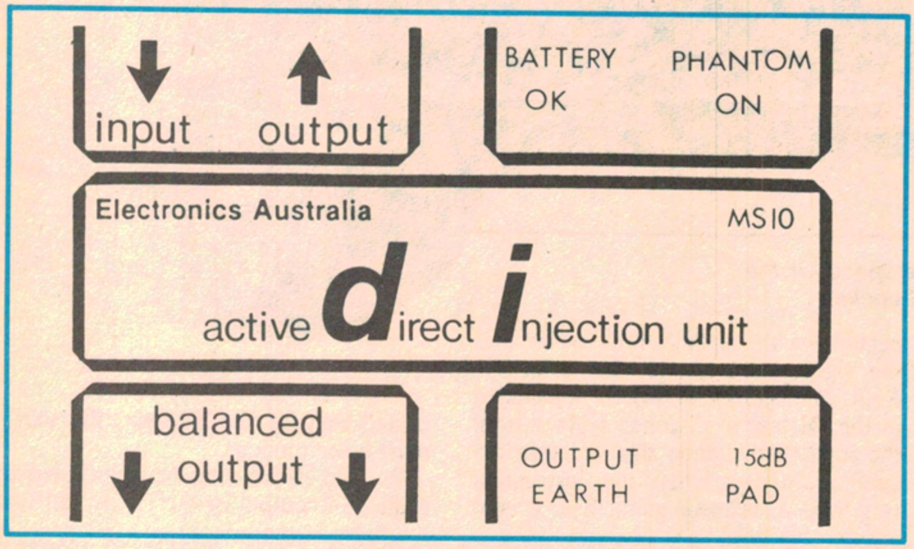
A "pad" switch has been included to reduce the output of the DI by 15dB so



The circuit uses a buffer and an inverter (IC1a and IC1b) to produce the balanced output, while the lower section of the diagram shows the power supply and voltage test circuitry.



The Scotchcal front panel was trimmed to allow easy access to the box lid screws, although the battery rarely requires replacement. The actual size artwork is shown below.



as to avoid overloading the input stage of the mixing desk, which is designed to accept microphone level signals. Only one level of attenuation is needed unless the DI is connected to the speaker output of a stage amplifier. Some DI's allow for this input level, but this style of connection is rather out of fashion, most engineers wanting the cleanest possible signal.

The average synthesiser can deliver up to a couple of volts of output when playing chords, and this large signal will produce some red lights and rather red faces at the mixing desk, unless some DI attenuation is available! Because of the limited supply voltage the DI itself can overload, but the prototype can take +15dBm before clipping; enough headroom for the nastiest synthesiser or pre-amp output.

Although the internal 9 volt battery will virtually last its shelf life, they have an embarrassing habit of giving up at the most inappropriate moment –leaving that inspired musical burst missed by the audience, or the tape machine. In this project the battery indicator LED1 will normally illuminate for a few seconds when the unit is switched on (plugging into the input socket). If the battery terminal voltage is below about 8 volts, the LED will not illuminate; the unit will still operate, but with a reduced headroom. If phantom power is present, the "phantom on" LED2 will also illuminate for a few seconds after the unit is turned on.

Phantom power

This is an expression traditionally applied to condenser microphones, where the power required for the internal pre-amp is supplied from the mixing desk via the balanced signal output lead. This is possible when the power is applied through suitable resistors to *both* pin 2 and pin 3. Since this voltage is common to both signal lines, it will be ignored by the audio signal processing system. A similar arrangement at the microphone end allows the pre-amp to draw a small current, pin 1 acting as the power earth as well as the audio shield.

Phantom power is becoming quite common in studios and sound reinforcement mixing desks, and only a few extra components were required to include this facility in our DI. Standard phantom power circuits (e.g., as per DIN 45596) are loosely adhered to by mixing desk manufacturers. Therefore our circuit components have been chosen for the DI to operate over a wide phantom power range, this voltage automatically taking over from the battery.

Circuit description

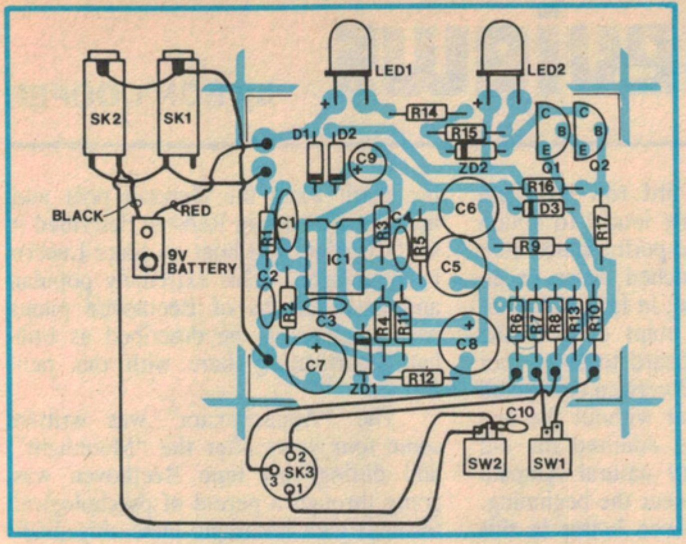
The basis of the circuit is a TL062 dual op-amp arranged as a buffer followed by an inverter, the two outputs supplying the in-phase and out-of-phase signals respectively. The TL062 was chosen for its high input resistance and extremely low power consumption: about 0.5mA from 9 volts. This leads to an excellent battery life.

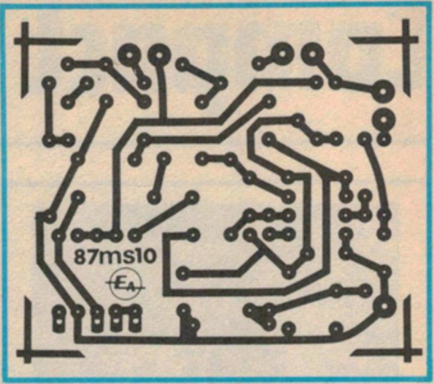
A slightly better noise figure could be obtained with a TL072 op-amp, but this would only be a few dB at the expense of a ten fold increase in supply current (5mA). In our lab, the prototype was measured at a distortion of 0.06% and noise of —86dB, both with respect to 0dBm.

An input at SK2 is applied to IC1a via a low pass filter, R1 and C2 (to remove any RF or other interference), and AC coupled by C1. R11 and R12 voltage divide the supply rail in half to provide a bias for IC1a and IC1b, via R2 and R4 respectively. The input is attenuated by about 6dB by R1 and R2, to compensate for the inherent 6dB gain when converting to a balanced system. If more output is required, the value of R2 can be increased. IC1 itself has a very large input resistance (1T ohm!).

The output of the buffer stage is fed to IC1b, which is arranged as an inverter with unity gain determined by R3 and R5. Once again, the high frequency response is limited to about 20kHz, in this case by C4.

Bipolar electrolytic capacitors are needed to couple the op-amps to the output, due to the variety of voltages that may occur in the output circuitry. The DI output has no DC component when operating from the internal battery, the 4.5 volt op-amp bias being decoupled by C5 and C6. When a phantom power voltage is present at pins 2 and 3 of SK3, the output capacitors have to cope with the opposite polarity of the previous situation. Normal elec-





Right: Full size PCB artwork (code 87ms10).

Left: Internal wiring and component overlay. Note C10 mounted on the terminals of SW2, and the solid wire used to connect the PCB to SW1.

trolytics would get quite upset!

Pins 2 and 3 of SK3 are supplied by R6 and R9, setting the nominal output impedance at about 600 ohms. R10 is connected across the output by SW1, the "pad" switch, reducing the output by 15dB for high level signal sources.

When phantom power is applied to pins 2 and 3, R7 and R8 couple this voltage via D2 to the op-amps supply pin (8). D1 is then reverse biased due to the higher phantom voltage, preventing current flow to the battery. Extra ripple filtering is provided by C8, while ZD1 prevents the supply rail rising above 12 volts.

According to phantom power standards, a 48 volt supply should have a source resistance of 3.4k ohms, and for 24 volts a 600 ohm source. It is debatable that these standards are always applied, but our circuit current of less than a milliamp allows a wide range of conditions to be corrected by ZD1.

The ring connection of the stereo input socket is wired to the negative battery terminal, and the earth lift switch. This enables the power circuit to be completed when an instrument is plugged in, via a mono jack lead. The input tip connection is linked to the output link socket, so the signal may also be sent to a stage amplifier for monitoring.

When power is applied to the DI, C9 will charge to the supply rail (D3 discharging it on power down), providing base current to Q2. This will cause the Darlington pair Q1 and Q2 to saturate for a few seconds, as set by R16, R17, and C9. If the supply voltage is greater than the breakdown voltage of ZD1 (5.6 volts), plus the forward voltage of LED1 or LED 2, current will flow.

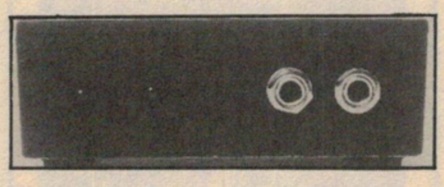
For a battery voltage of 9 volts, about 5mA will flow (as set by R14); illuminating LED1. The current through LED2 is mainly set by R7 and R8, in conjunction with the resistance and voltage of the phantom source.

The output earth switch SW2 disconnects the DI ground from pin 1 of the XLR socket SK3, C10 providing RF continuity. Although SW1 breaks the shield connection to the mixing desk, the phantom power still has a voltage return path via the mains earth (its presence causing the original earth loop!).

Construction

Start the construction by assembling all of the smaller components on the PCB (code 87ms10), paying particular attention to the orientation of the semiconductors and electrolytic capacitors. The two LEDs are soldered to the PCB with about 15mm legs, which are bent at right angles (in the middle). Because the assembled PCB is quite small and light in weight, the mounted LEDs will easily support one side. The other side is supported by short lengths of solid wire (component leg offcuts about 15mm long), soldered between the "pad" connections on the PCB, and the "pad" switch.

Next, solder lengths of lightweight hookup wire to all the input and output points on the PCB, as indicated in the component overlay. These will later be wired to the sockets, switches and the battery terminal.



DI side view, showing the instrument in and out sockets and the power test LEDs.

The box should now be prepared by drilling (and filing) the appropriate size holes for the sockets, switches, and LEDs. Position these holes in the centre of the box height, using the pictures to get an idea of the layout along the sides. This positioning is not critical, but make sure a space is left for the battery to fit snugly between the PCB and the 6.5mm output socket.

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PARTS LIST

1 diecast box, 65x40x120mm
1 PCB,code 87ms10, 40x50mm
1 male 3 pin XLR panel mount socket
2 6.5mm stereo jack sockets (enclosed type)
2 miniature SPST rocker switches

Semiconductors
1 TL062 dual op-amp
2 BC547 NPN transistors
1 5.6 volt 1 watt zener diode
1 12 volt 1 watt zener diode
3 1N4148 diodes
2 5mm LEDs

Capacitors
2 47uF 25VW PCB mount electrolytics
1 1uF 50VW PCB mount electrolytic
2 4.7uF 50VW bipolar electrolytics
3 10nF greencaps
1 1nF greencap
1 15pF ceramic

Resistors (all 0.25W 5%)
1 x 100Ω, 1 x 220Ω, 1 x 270Ω, 2 x 560Ω, 1 x 2.2k ohm, 4 x 4.7kΩ, 3 x 100kΩ, 2 x 470kΩ, 2 x 1.2MΩ

Miscellaneous
9 volt battery and suitable snap connector, rubber feet, nuts and bolts, 2 x LED mounting kits, hookup wire.

DI BOX

Continued from page 67

The prototype case was rubbed down with a fine sandpaper, sprayed black, and a Scotchcal front panel attached. The sockets and switches may now be mounted and the circuit board installed. After mounting the LEDs in the mounting kits, the short lengths of solid wire can be soldered firmly to the Pad switch, thereby securing the PCB.

By referring to the component overlay, the hookup wire from the PCB can be cut to length and soldered to the appropriate sockets and switches. Don't forget the wires for the output earth switch, or the small capacitor mounted across its terminals. If it is difficult reaching some of the socket and switch connections with a soldering iron, they may need to be temporarily removed to attach the wires.

If the battery is still a little loose when installed, it may be held in place by a small piece of foam or polystyrene. This can be placed under the lid, before it is screwed down.

Using the DI

Connecting the DI to a system is very simple; the instrument or signal source plugs into the input, and the output is fed to a stage amplifier if used. Connect the balanced XLR output as if a microphone was the signal source, that is, plug into a microphone lead that feeds the input of a mixing desk.

Some interpretation of the battery indicator LED is possible. Although it will not illuminate if the battery is below about 8 volts, it gets quite dull as the voltage approaches this point.

The phantom indicator will light only if phantom power has been applied (via the XLR connections), *and then* the DI is switched on (by plugging into the input).

The output earth switch normally should be left in the "on" position, and switched off if an earth hum loop is present. When only an instrument (i.e. no stage amplifier) is used with the DI box, the earth switch should always be in the "on" position.

The pad switch will most likely be "on" for signal sources such as keyboards, pre-amp outputs, and high output bass guitars. The "off" position will probably apply to instruments such as acoustic guitars (with a "bug"), and normal bass guitars.

Our tests have shown that this DI is capable of very high quality results. Experience will show the best way to use it.