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EDITION 1

Assembly Manual

Speech Processor for Transceivers

K-6002

DICK SMITH ELECTRONICS

ACN 000 445 956

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Increase your 'talk power' with this simple but effective little unit. It connects in-line with your existing microphone, and acts as a preamplifier, compressor and speech filter. As a bonus, it also automatically generates a 'beep' each time the PTT button is released, to signal the end of your 'over'.

If you've ever watched a level meter that's responding to a human voice, you may have noticed just how much higher the peaks are than the average speech level. This effect is particularly troublesome when a voice is used to modulate the carrier in a radio transmitter, where the peaks can cause over-modulation and frequency 'splatter'.

To avoid this problem, the operator usually adjusts the microphone gain control to a point where the speech peaks are only *just* providing 100% modulation.

In this case however, the soft sounds are often barely heard at the receiving end, despite the clean nature of the signal. In short, the problem lies with the ratio between the loud and soft components (or dynamic range) of typical human speech.

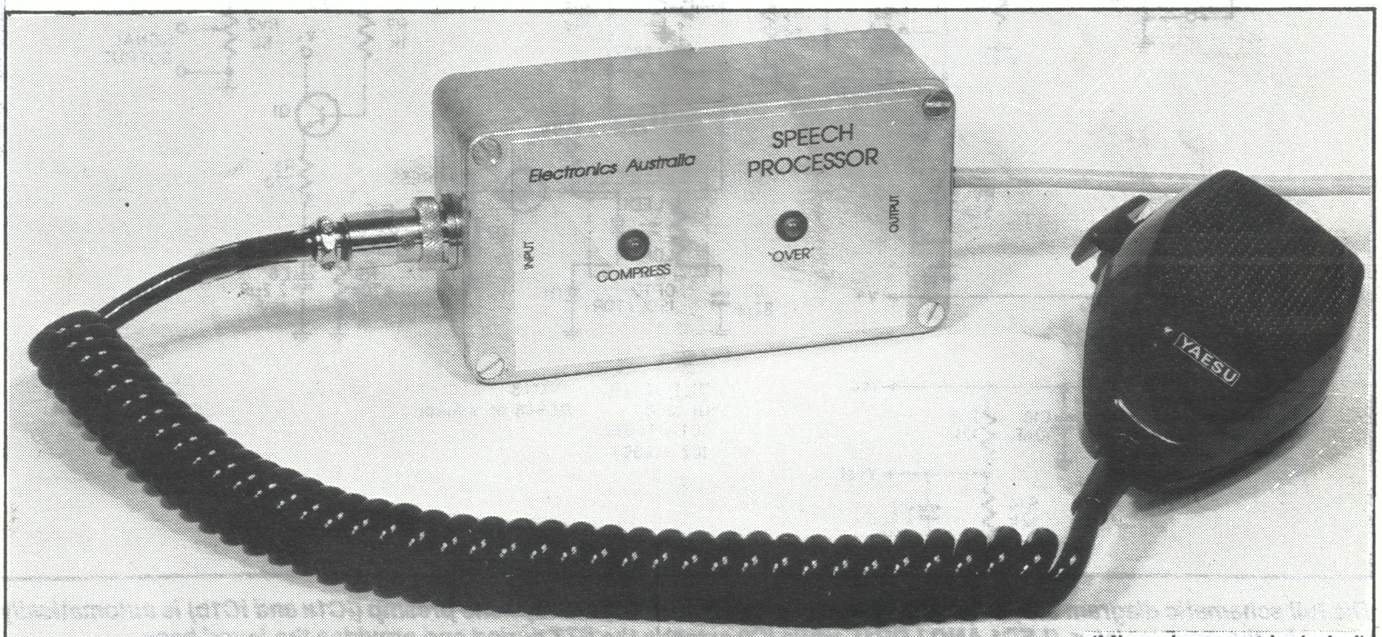
The simplest way of improving this situation is to pass the audio signal through some form of hard-limiting circuit, before it's applied to the transmitter. In this way, the mic gain can be increased for a stronger signal at the receiving end, and when the mic signal exceeds a preset volume level, the waveform is simply 'chopped off'. When clipping does occur however, the resulting speech signal is rather unintelligible, and contains a large number of high-order harmonics which are difficult to filter out — so this solution is less than satisfactory.

A more successful method of controlling the dynamic range is to use some form of *compressor*, or ALC (automatic level control) circuit.

With this arrangement, the maximum signal level is restricted by a special amplifier stage, which is able to reduce its gain in response to large signal peaks. The transceiver's mic gain can then be set to a higher level, and the average level of modulation increased by a substantial degree.

With this in mind, we have developed a simple circuit to perform the required compressing/limiting function, and included a number of other features to make the unit even more useful.

The final design presented here uses



(Microphone not included)

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common low-cost components, and results in a substantial increase in a transmitter's overall communication efficiency. The main extra feature to be included in the speech processor is a circuit which delivers a short audio tone (indicating 'over' or 'Roger') at the end of each transmitting period — or in fact, as the transceiver's press-to-talk (PTT) button is released. To allow the tone to pass through the activated transmitter, the PTT action needs to be slightly delayed past the point where the operator actually releases the button — this has been achieved by adding just one CMOS chip, a relay and a few extra components.

Other than that, we've included substantial audio filtering through the circuit, so as to restrict the signal's energy to the nominal 300Hz to 3kHz speech bandwidth. Again, this improves intelligibility at the receiving end and makes the most efficient use of the available transmitter power.

The circuit itself can be powered from any 12 volt (or thereabouts) DC power source, and is housed in a standard diecast aluminium box to provide RF shielding. Also, indicator LEDs have been included to show the operator when the compression and 'over' circuits are operating.

Circuit description

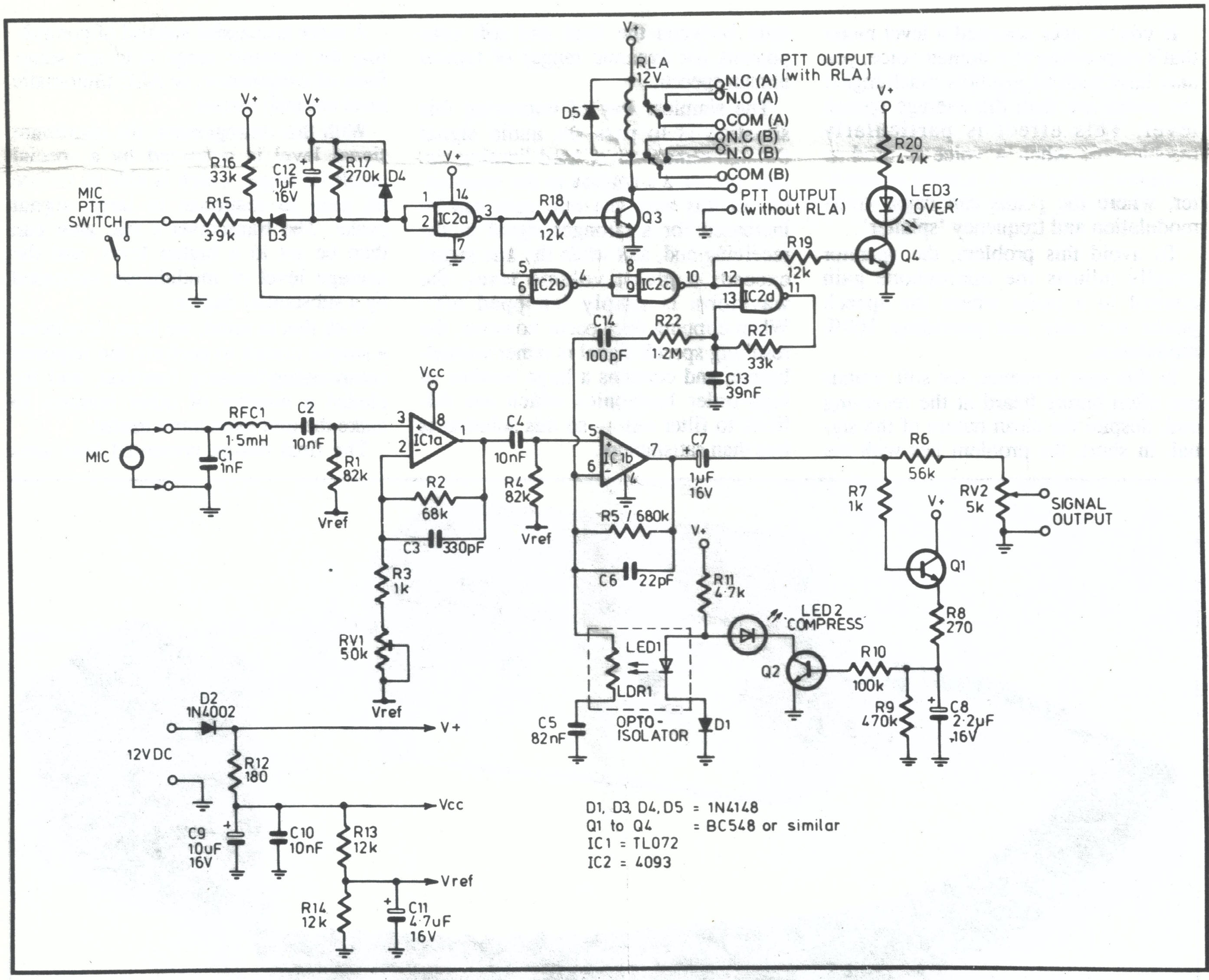
The circuit of the Speech Processor can be divided into two main areas: the delayed-PTT section based around IC2 as shown in the top half of the schematic diagram, and the audio preamplifier/compressor formed by IC1 and its associated components in the lower half of the diagram. Both parts of the circuit operate in quite a straightforward manner, and should be quite easy to faultfind — should the need ever arise.

The low-level microphone signal is applied to the preamp/compressor section at C1 (which provides a low impedance path to any stray RF energy), and via RFC1 (which effectively blocks any remaining high-frequency signals) to the high-pass filter composed of R1 and C2. This filter rolls off the unwanted low-frequency content of the incoming speech signal, and provides a DC biasing path to Vref (around 6V) for IC1a.

IC1a is configured as a standard non-inverting amplifier stage, with the gain set by the combination of R2, R3 and RV1 — this will be anywhere between 2.3 and 69, depending upon the setting of RV1. The overall gain of the stage is reduced at high frequencies by the feedback action of C3, which helps to reduce the unwanted higher-order harmonics of the speech signal.

The output at pin 1 is then transferred to the following non-inverting amplifier (IC1b), via a further high-pass filter composed of C4 and R4; as before, the input is biased to Vref via R4. This second stage forms the compressor part of the circuit, with the gain (mainly) set by components R5, LDR1 and C5.

Assuming for the moment that LDR1 has a resistance of about 15k (more of this later), the stage will have a gain of about 45 at the middle speech frequencies (say 1kHz) since C5 will offer a relatively low impedance under these conditions. At very low frequencies on the other hand (say below 200Hz), C5 will present a much higher impedance and the stage gain will be substantially lower. So in effect, this is yet a further



The full schematic diagram of the Speech Processor. The gain of the microphone preamp (IC1a and IC1b) is automatically varied by the opto-isolator (LED1 AND LDR1), while IC2 extends the PTT period and provides the 'over' beep.

Text and illustrations courtesy of Electronics Australia

high-pass filtering action to reduce the audio bandwidth. Also, as in the previous stage the higher frequencies are rolled off by the effect of C6.

The combination of all of these cascaded low and high-pass filters reduces the processor's overall frequency response to our desired range of around 300Hz to 3kHz, with reasonably steep cut-off slopes at either end. The amplified and filtered signal at the output of IC1b (pin 7) is then passed to the processor output via the coupling capacitor C7, and an output attenuator composed of R6 and RV2. This reduces the processor's output signal a suitable level for the normal microphone input of a transceiver.

Compression

The compression action of the processor is controlled by the circuit based around Q1, Q2 and the linear opto-isolator formed by LDR1 and LED1.

Since the gain of the second preamp stage (IC1b) is dependent upon the resistance of LDR1, which in turn depends on level of light transmitted to its surface by LED1, we can easily control the processor's output signal level by varying the current through the LED section of the opto-isolator.

With the circuit components as shown, a standing current of around 2mA flows through LED1 (as set by R11), which ultimately sets LDR1 to a resistance of about 15k, and the preamp stage to a gain of 45.

However, if transistor Q2 is turned hard on, the current from R11 will be shunted to ground via LED2 and Q2's collector, rather than via LED1 and D1 — this occurs since Q2 will now represent a much lower resistance than D1.

As you would expect, LED1 will now extinguish, LDR1 changes to a very high resistance (many megohms), and the stage gain drops to around unity. The processor's overall gain has now dropped by a substantial degree, and LED2 is illuminated to indicate this condition — that is, that the unit is compressing.

In practice, the circuit uses Q1 to continuously sense the output signal level, and ultimately change the bias on Q2 (and the circuit gain) accordingly.

Incidentally you may be wondering why have we used a 'home-made' opto-isolator to perform this gain control function, instead of a standard 'ready made' opto-coupler like the familiar 4N28. The short answer is that standard couplers are essentially digital devices, and not suitable for the kind of linear control we need here.

A simple LED/LDR combination as used here gives the right control characteristic, at very low cost.

The processor's high-level output signal is passed via R7 to Q1, which ostensibly acts as an emitter follower for any signal more positive than its emitter potential. Therefore, as the output signal swings below ground potential (the negative half of the cycle) Q1 is reverse biased, while C8 is

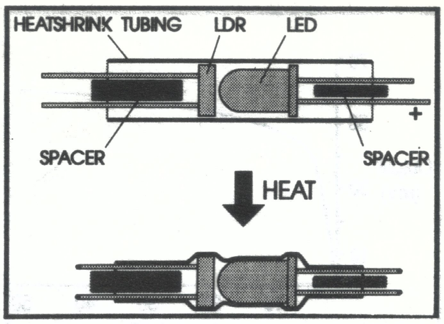


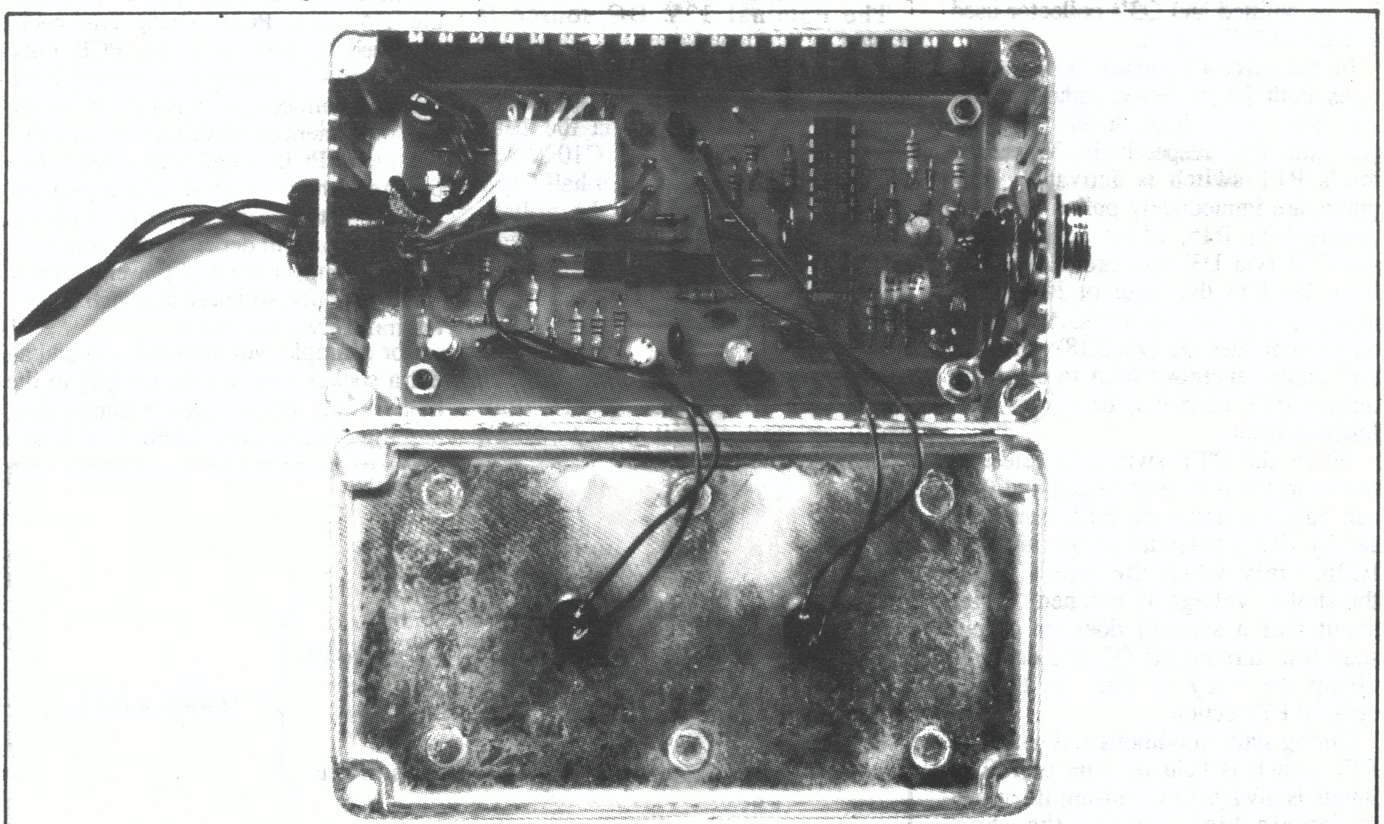
Fig.1: The construction technique for the opto-isolator. Simply slide the components into the heatshrink tubing and apply heat.

charged via R8 as the output swings in a positive direction.

In this manner, C8 will charge to the peak voltage level of the output signal's positive excursions (less the 0.6V drop across Q1's base/emitter junction). Since the voltage across C8 sets Q1's emitter potential, the transistor will remain reverse biased until C8 begins to discharge (via R9 and R10) or the signal level increases. So in effect, this part of the circuit behaves as a positive peak rectifier, with a buffered input.

The resulting DC signal then provides the bias current for Q2, which as mentioned above, ultimately sets the overall gain and output signal level of the processor.

In this way a gain control loop is established only when the signal at R7 exceeds a peak level of about 1.2 volts, where both Q1 and Q2 are begin-



An inside view of the completed processor. Note that a couple of components (namely RFC1 and R15) are connected between the PCB and the microphone input socket, and the grounding wire is terminated at one of the PCB mounting bolts.

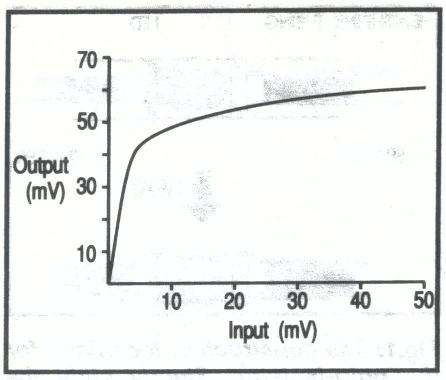


Fig.2: The audio compression curve for the processor's preamp stage. While the curve's shape remains the same, the values along each scale will depend on the settings of RV1 and RV2.

ning to conduct and the circuit gain is forced to drop.

The response time of the DC control signal at C8 is tailored by the circuit components R8 and R9, which set the capacitor's charging (attack) and discharging (release) rates respectively.

'Roger' overtones

The circuit shown in the upper half of the schematic diagram uses RLA's contacts to repeat the action of the mic's PTT function, but with slight additional delay in the releasing action to allow the 'over' tone to be transmitted. If the transceiver's PTT action can be toggled by a simple grounding switch (that is, connecting the PTT line to 0V), RLA can be omitted and Q3's collector used as a direct PTT switch.

In the circuit's steady-state conditions, both the anode and cathode of D3 are held at a high level (V+) by R17 and R16 respectively. When the mic's PTT switch is activated, both points are immediately pulled to a low potential via R15, which rapidly charges C12 (via D3) and provides a low logic level to the input of IC2a. The resulting high level at the NAND gate's output activates Q3 (via R18) which in turn either energises RLA to close the output PTT contacts, or serves this function itself.

When the PTT switch is released however, D3 is reverse biased and C12 can only discharge via R17, causing a slowly rising potential at the input of IC2a. Only when the gate's input threshold voltage is reached (after about half a second) does its output snap low, turning off Q3 and de-energising the relay — thus we have a delayed PTT action.

During static conditions and when the PTT switch is held on, one of IC2b's inputs is always low, causing its output to remain high. During the short 'extension' of the PTT period however, both inputs (pins 5 and 6) are momentarily high, forcing the output of this gate to go low for around half a second. This level is inverted by IC2c and applied to one input of IC2d (pin 12) as a high-going pulse. IC1d is arranged as a simple gated oscillator, with a frequency set to about 500Hz by the combination of C13 and R21. To produce the final 'over' tone, we have elected to use the triangle waveform appearing at pin 13 of IC2d, rather than the usual squarewave signal (at pin 11) with its excessive upper harmonics.

When pin 12 pulses high, the tone is coupled to the second mic preamp stage via R22 and C14, which attenuate and filter the waveform by a large degree. In fact, most of the attenuation is performed by C14 — which at the same time minimises the effect of the DC shift which occurs as the oscillator starts.

As an indication of the 'over' tone period, the positive-going pulse at pin 10 of IC2c forward biases Q4 via R19, which in turn illuminates LED3 via the current limiting resistor R20. The LED current has been restricted to a modest 2mA level so as to match the brightness of the 'compress' indicator LED2.

By the way, D4 and D5 have been included as component protection diodes — D4 offers a discharge path for C12 when the power supply is disconnected, while D5 suppresses the backswing voltage generated by RLA's coil when Q3 turns off.

The final small section of the schematic produces the power supply rails for both the analog and digital parts of the processor's circuit.

The nominal 12V DC source is passed to the majority of the circuit via the protection diode D2 (V+), while the more sensitive preamp section based around IC1 is supplied via the filter components R12, C9 and C10 (Vcc). Finally, a reference supply of half Vcc's potential is generated by the voltage divider R13 and R14, and then filtered by C11 (Vref).

Construction

The majority of the Speech Processor's components mount on one small printed circuit board (PCB), measuring 102 x 56mm and coded 91ap8, which neatly fits into a standard 120 x 40 x 65mm aluminium diecast box. The microphone connects to a matching socket at one end of the box, while the processed signals are passed on to the tranceiver via a length of four-core shielded cable, which is terminated in an appropriate mic plug.

Start the construction by assembling the opto-isolator as shown in Fig.1. First, slide the LED and LDR into a short length of 6mm heatshrink tubing so that their surfaces are just touching, and insert a couple of small spacers to prevent component legs from shorting together — short pieces of insulated hookup wire make suitable spacers.

Then apply heat to the heatshrink tubing, while making sure that the spacers remain between the legs of the components. The completed opto-isolator should then be tested with a multimeter to check that the LDR is at a very high resistance (since no light should be falling on its surface), and to determine which is the positive leg of the LED.

The isolator's four legs can now be bent to suit the layout of the PCB.

The circuit board can now be loaded with all of the components, including the opto-isolator. As usual, take particular care with the orientation of the polarised parts such as electrolytic capacitors and semiconductors — refer to the component overlay at all times.

The DSE K-6002 kit is supplied with a D.P.D.T relay and this fits straight into the ZA 1396 PCB which has been modified to suit the relay. PCB pins should now be added to the board at all external connection pads (as shown in the component overlay diagram), and the completed PCB assembly attached to the bottom of the box with small screws, nuts and insulated spacers. The final wiring arrangement will depend upon how you plan to connect the 12V power source - which of course will need to match that of the transceiver mic input.

For example, you may wish to add an extra socket (say a 3.5mm-type) to the box for the power supply connection, and run the four-core shielded cable to a

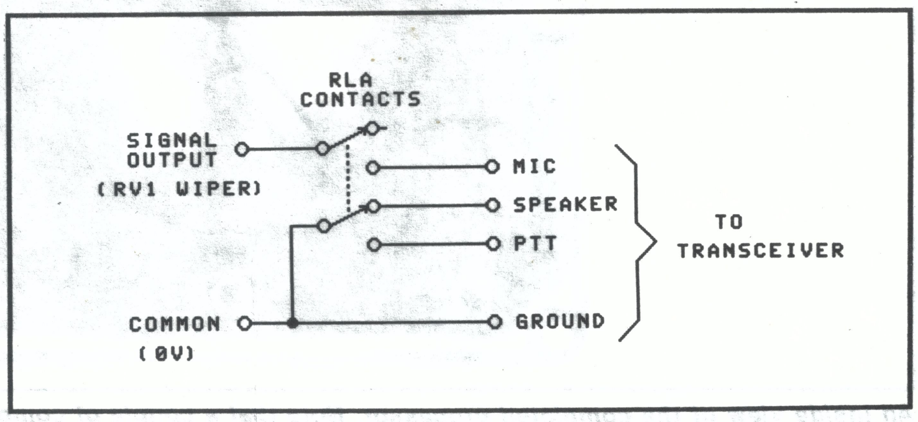
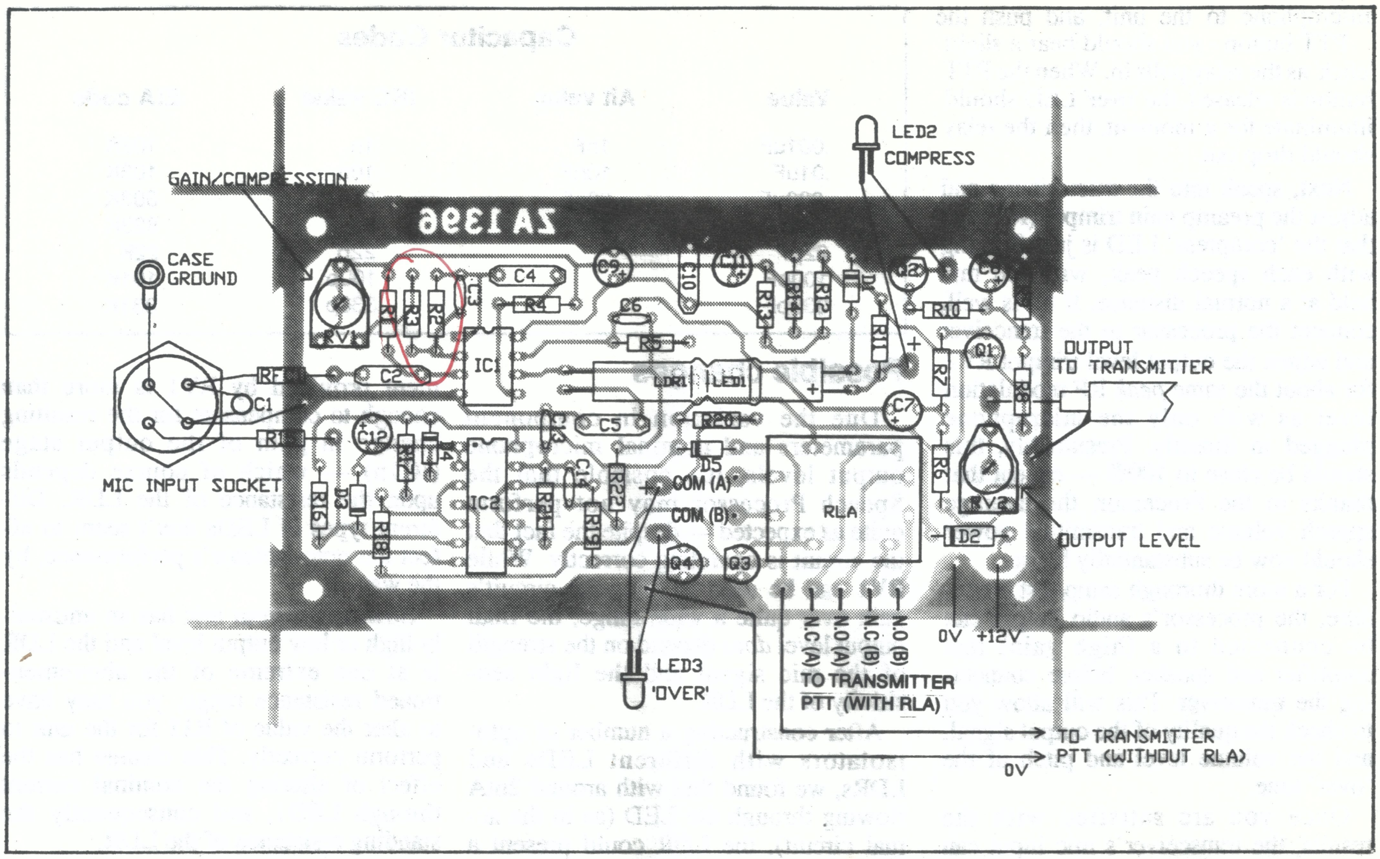


Fig.3: A suggested wiring arrangement to suit some tranceivers (usually CB) which use a DPDT switch in their matching microphone unit.



Refer to this component overlay and wiring diagram during the construction of the Speech Processor — pay particular attention to the orientation of polarised components.

PARTS LIST

Resistors (All 0.25W)

- R1 82K (gry-red-org)
- R2 68K (blu-gry-org)
- R3 1K (brn-blk-red)
- R4 82K (gry-red-org)
- R5 680K (blu-gry-yel)
- R6 56K (grn-blu-org)
- R7 1K (brn-blk-red)
- R8 270R (red-vio-brn)
- R9 470K (yel-vio-yel)
- R10 100K (brn-blk-yel)
- R11 4.7K (yel-vio-red)
- R12 180R (brn-gry-brn)
- R13 12K (brn-red-org)
- R14 12K (brn-red-org)
- R15 3.9K (org-wht-red)
- R16 33K (org-org-org)
- R17 270K (red-vio-yel)
- R18 12K (brn-red-org)
- R19 12K (brn-red-org)
- R20 4.7K (yel-vio-red)
- R21 33K (org-org-org)
- R22 1.2 Meg (brn-red-grn)
- RV1 50K 5mm horz. Trimpot
- RV2 5K 5mm horz. Trimpot

Capacitors

- C1 .001uF Ceramic
- C2 .01uF Greencap
- C3 330pF Ceramic
- C4 .01uF Greencap
- C5 .082uF Greencap
- C6 22pF Ceramic
- C7 1uF 25/50V RB Electro
- C8 2.2uF 25/50V RB Electro
- C9 10uF 16/25V RB
- C10 .01uF Greencap
- C11 4.7uF 25/50V RB Electro
- C12 1uF 25/50V RB Electro
- C13 .039uF Greencap
- C14 100pF Ceramic

Semiconductors

- Q1 BC/DS 548 Transistors NPN
- Q2 BC/DS 548 Transistors NPN
- Q3 BC/DS 548 Transistors NPN
- Q4 BC/DS 548 Transistors NPN
- D1 1N914/1N4148Diode
- D2 1N4002/4 Diode
- D3 1N914/1N4148 Diode
- D4 1N914/1N4148 Diode
- D5 1N914/1N4148 Diode
- Led1 5mm Red Led
- Led2 5mm Red Led
- Led3 5mm Red Led
- LDR1 LDR OP12/DSC DS01
- IC1 TL 072CP/LF 353 IC
- IC2 4093 IC

Miscellaneous

1 x PCB, coded ZA 1396, 1 x 120x40x65mm diecast aluminium box, 1 x front panel label, 1 x 1.5mH RF choke (RFC1), 1 x DPDT relay (RLA), 1 x plug mic 4 pin socket, LED mounting hardware, heatshrink tubing, cord clamp, solder lug, four core shielded cable, hookup wire, solder, PCB pins, screws, nuts and washers.

microphone plug which matches the tranceiver socket. Alternatively, a reasonable length of two-core cable could be run out of the box for a direct connection to the power source.

Note that components RFC1 and R15 are wired between the PCB and the input socket, rather than mounted on the board itself, and capacitor C1 is connected directly between the input pins on the mic socket. The RF choke should be mounted as close to the input pins as possible for maximum RF rejection, and a grounding wire connected between the input socket and the case (as shown in the overlay diagram). Finally, the two LEDs can be installed in the front panel, and wired to the appropriate PCB pins.

By the way, many CB transceivers use a double pole switch inside their matching microphone to perform the PTT changeover action. The switch is wired so that the transceiver's speaker is disconnected during 'transmit' and the microphone is disconnected during 'receive', as well as providing the usual PTT line which activates the transmitter.

In this case the relay's double-pole changeover contacts can be wired so as to mimic the function. See Figs.3, 4 & 5 for typical wiring arrangements.

Setting it up

The initial adjustments and tests should be completed before connecting the Speech Processor to a tranceiver. First, connect the power supply and microphone to the unit, and push the PTT button - you should hear a slight 'click' as the relay pulls in. When the PTT button is released, the 'over' LED should illuminate for a moment, then the relay should drop out.

Next, speak into the microphone and adjust the preamp gain trimpot (RV1) so that the 'compress' LED is just flashing with each speech peak, with the mic held at a normal distance. If all is well, connect the processor to the tranceiver and adjust the output level trimpot RV2 for about the same *peak* RF modulation level as with only the microphone plugged in directly. Presumably, this should be close to 100% — except that thanks to the Processor, the average speech volume and transmitted power should now be substantially higher.

For a more thorough setup/test procedure, the processor's audio output can be connected to a (high gain) test amplifier and speaker, before connecting the tranceiver. This will allow you to check the quality of the output signal, and the volume level and pitch of the 'over' tone.

Once you are satisfied with the results, the transceiver's mic input can be connected and the processor output level set as mentioned above.

Capacitor Codes

Value	Alt value	IEC value	EIA code
.001uF	1nF	1n	102K
.01uF	10nF	10n	103K
.039uF	39nF	39n	393K
.082uF	82nF	82n	823K
22pF		22p	22K
100pF		100p	101K
330pF		330p	331K

Possible changes

Due the variation in component parameters and nominal microphone output levels, it's possible that the Speech Processor may not perform quite as expected — despite the fact that the circuit is operating correctly. While RV1 can be used to alter the circuit's gain over quite a wide range, the final output level *does* depend on the strength of the mic signal and the light sensitivity of the LDR.

After constructing a number of opto-isolators with different LEDs and LDRs, we found that with around 2mA flowing through the LED (as in the actual circuit), the LDR could present a resistance anywhere between 10k and 50k. As it happens, the range of adjustment provided by RV1 is more than enough to compensate for the resulting change in gain of the output stage (IC1b) — which of course depends upon the resistance of the LDR. Different types of LEDs don't seem to effect the opto-isolator's performance, by the way.

However, if your mic has an unusually high or low output level and the LDR is at one extreme of the abovementioned resistance range, you may have to alter the value of R11 for the unit to perform correctly. This change has the effect of altering the nominal current through LED1, and consequently the standing resistance of the LDR.

Some constructors may wish to test the opto-isolator *before* it is installed in

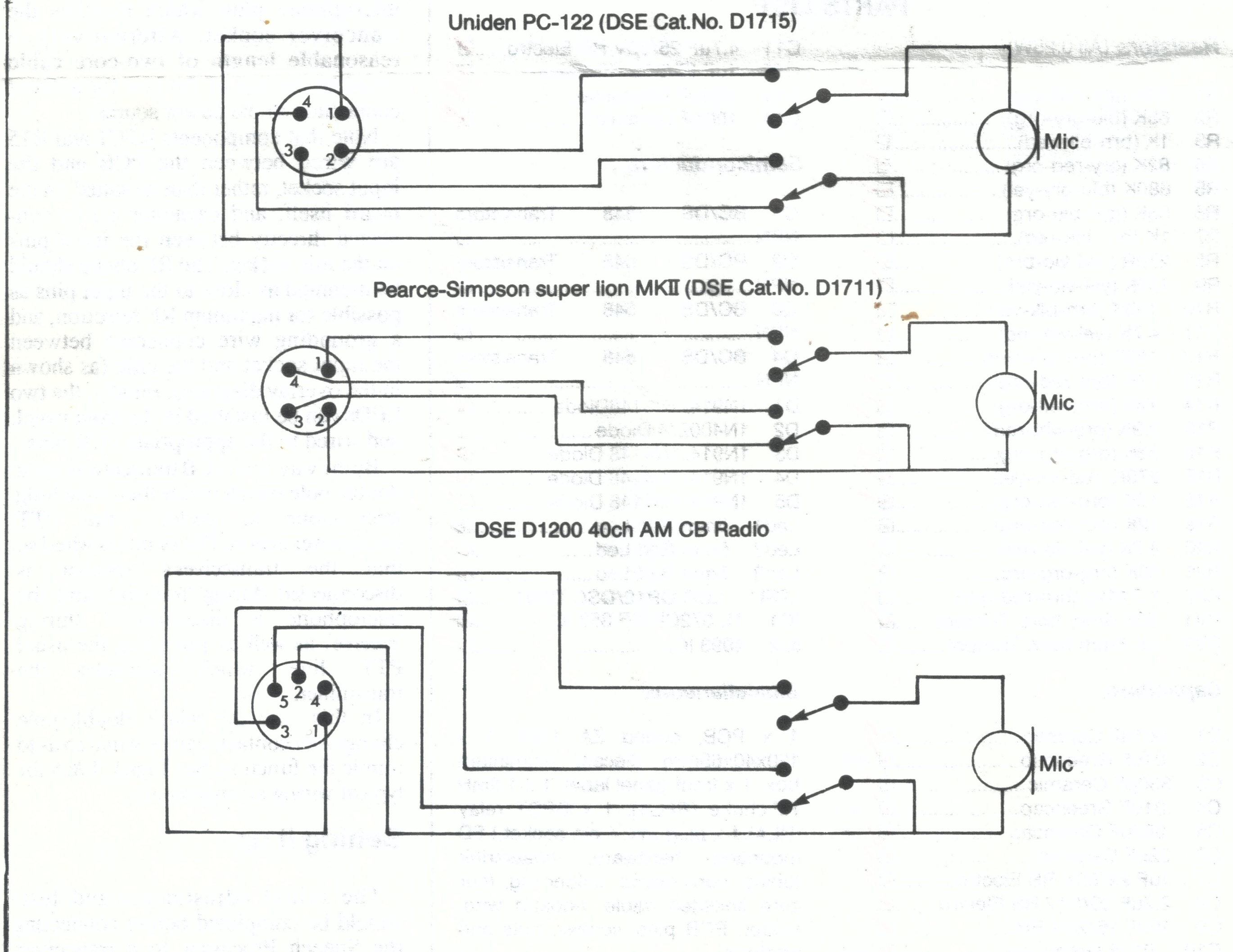


Fig. 4: Common CB radio P.T.T diagrams.

the PCB. In this case, simply measure the resistance of the LDR, while changing the current through the LED. When you find what LED current corresponds to an LDR resistance of around 15k (a little higher is fine), calculate the appropriate value for R11 and assemble the circuit board with these components installed.

The other possible anomaly with the processor's operation concerns the duration and pitch of the 'over' beep. Since the timing of these functions is dependent upon the input voltage at which the NAND gates change state (as well as the surrounding component values), any difference in this threshold level will have a marked effect.

We've found in the past that the input threshold of 4093 CMOS chips can vary quite significantly, depending upon the manufacturer. In this case, the answer is to vary the value of the associated components to achieve the correct timing periods — this will be R17 for the 'over' tone duration, and C13 or R21 for the corresponding pitch.

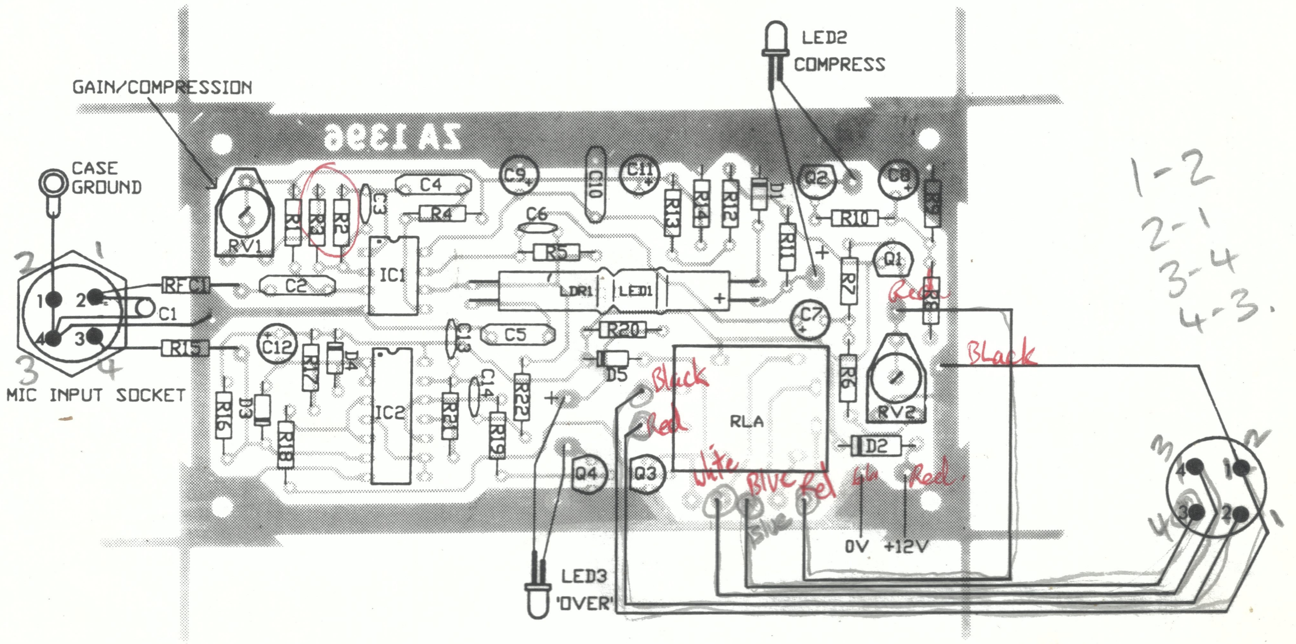


Fig.5: Example showing the speech processor connected to the uniden PC 122 Transceiver.

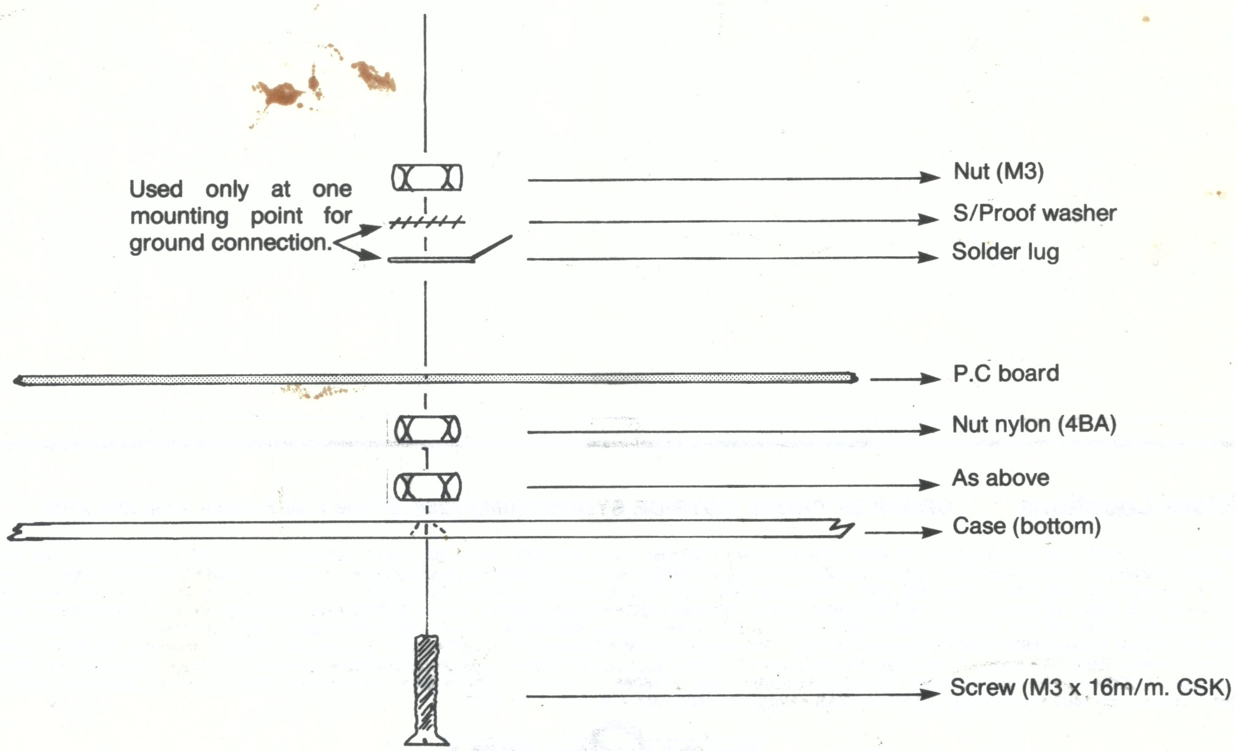


Fig.6: Mounting of the P.C board.

Notes & Erratas

Constructors please note: If you would like to remove the 'beep' from each time the PTT button is released (to signal the end of your 'over'), you can do so by shorting out C13 (39nF) capacitor. A SPDT Toggle switch could be used so that you could still use both options.

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