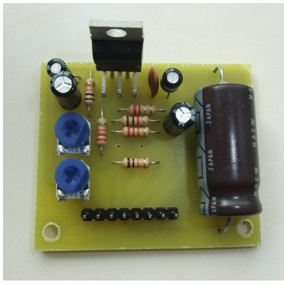
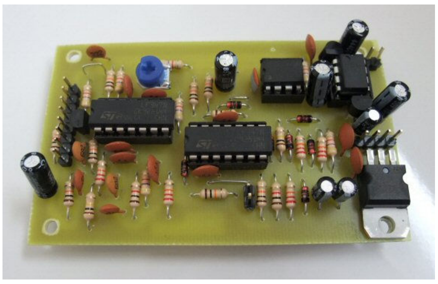
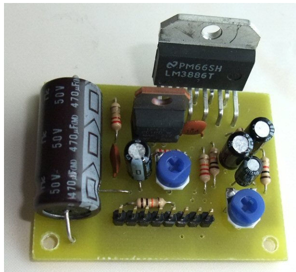
FAT5 Modulator
Information Manual
VERSION 1.0.1







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1. Introduction

Transmit audio has been the subject of much invention and discussion since the early days of AM broadcasting, and articles such as http://www.thebdr.net/articles/audio/proc/proc-hist.pdf provide a fascinating insight into the broadcasting industry's treatment of audio over the years. Amateur AM transmitters however have generally used badly designed modulators, most consisting of a crude audio amplifier and little else. There is no audio processing other than an LF roll-off, designed mainly to enable poorly regulated power supplies and smaller modulation transformers to be used. With no peak limiting, modulators with enough power (not always a given) will inevitably over-modulate the carrier. This historically is the preferred method for making the signal 'louder', but with SDR receivers coming of age the true effect of this approach is now all too readily visible. Amateur AM needs to clean up its act.

The FAT5 modulator is part of the FAT5 project, which is a Class E series-modulated AM transmitter for 80m and other frequencies. The modulator consists of a speech processor PCB and a power modulator PCB using external power transistors. There is a choice of two power modulator designs which cater for low/medium and medium/high power levels.

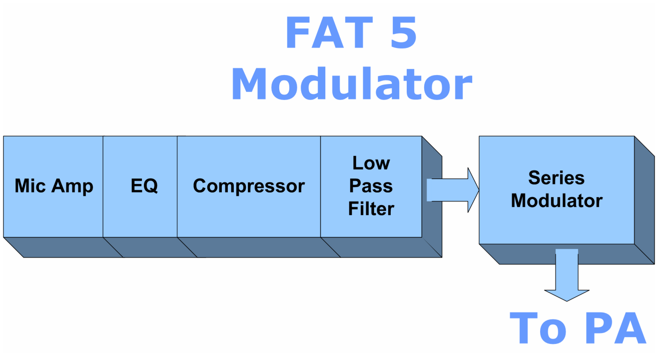

Figure 1.1 Block diagram of the FAT5 Modulator

1.1 Speech processor (FAT-MAX)

The FAT5 audio processor makes no claim to perfection, but it *will* provide a louder sounding signal than unprocessed audio and it *will* produce a well controlled modulation envelope. Figure 1.1 above shows a block diagram of the FAT5 modulator. The speech processor PCB handles all the low level audio processing, mic amp, EQ (equalisation), compression and bandwidth limiting.

Speech compression by James L. Tonne W4ENE
The speech compression part of the circuit is the design of James L. Tonne, W4ENE (ex WB6BLD), and the author is grateful to James for his support and valuable insight into the processing of audio. His excellent compression circuit is reproduced unaltered. The pre-compression mic amp and EQ stages are also inspired by James' filter circuits although many component values have been changed. Readers may know of James Tonne from his popular 'Elsie' LC filter design program.

Visit http://tonnesoftware.com/appnotes/speech/speechamp.html for an excellent paper on audio processing. Go to the home page for much more information and free software.

1.2 Series Modulator

The FAT5 series modulator stage is a Class-A amplifier consisting of a power amplifier chip driving a number of current amplifier transistors, the emitter load of which is the RF PA. Driving the modulator is a speech processor contained on its own PCB for use on other projects. The speech processor takes a mic level signal, amplifies it, shapes the frequency response, applies compression for improved talk-power then filters the signal through an 8-pole low-pass filter to tightly control the transmitted bandwidth.

The amplifier chip (TDA2050 or LM3886) is biased for single rail operation, so the resting voltage on its output pin is half the supply voltage. Audio passed through the amplifier is able to vary the output voltage from zero to full HT (ideally) thus producing 100% modulation of the RF stage.

Being Class-A, modulator efficiency is poor and adequate heat-sinking must be provided for the power transistors. Three TO3 cased power transistors are recommended for the 50W version, mounted on a suitable heatsink (which can be made substantially smaller by fan-cooling).

There are more efficient modulation methods including Class H and Class G that use split power supplies such that higher voltages are only applied to the modulating transistors on the positive half of the waveform thus greatly reducing heat losses. A circuit may be described in future versions of this manual however it is debatable whether the extra complication is worth the trouble at these power levels. A Class-D modulator using Pulse Width Modulation (PWM) is also under development, again more as a technical exercise.

The type of modulator used makes the power supply requirements less stringent. As long as the supply to the amplifier chip is stabilised, any ripple on the collectors of the main modulating transistors has minimal effect (as long as the power supply can meet the voltage demand at all current levels). In this sense the modulator can be thought of as a variable voltage series-pass stabilised power supply.

One may ask what the point is of a very efficient (85%+) Class E RF PA when the modulator is by comparison so inefficient but every Watt of heat saved in the PA through greater efficiency is a Watt also saved in the modulator as they are in series and the common point is at half HT volts. (In fact the modulator consumes slightly more Watts due to circuit losses such as base/emitter voltage and emitter balancing resistor dissipation)

2. FAT-MAX SPEECH PROCESSOR

The speech processor is built on a small PCB approximately 50mm x 80mm, as pictured below. The kit of parts contains all PCB mounted components. There are four mounting holes, one at each corner, drilled for M3 screws. The mounting hole under VR3 aligns with the tab on VR3 to allow extra heatsinking to be added if the regulator is also supplying an external load.

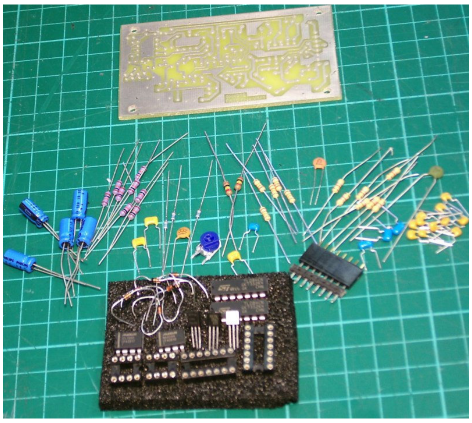

Figure 2.1 FAT-MAX kit of parts

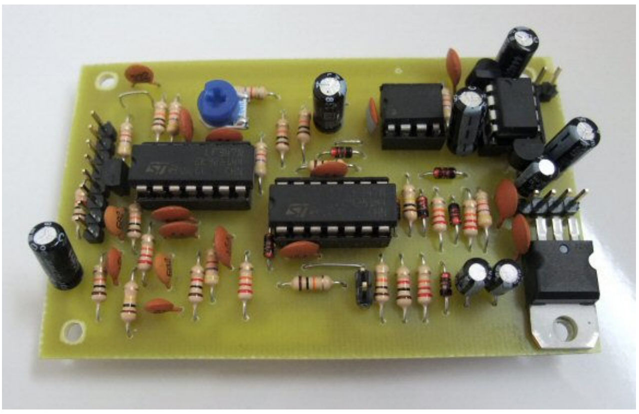

Figure 2.2 Assembled processor kit

The circuit can be broken into four separate areas. Each is explained in turn. The complete circuit is shown in figure 2.8.

2.1 Mic amp and EQ

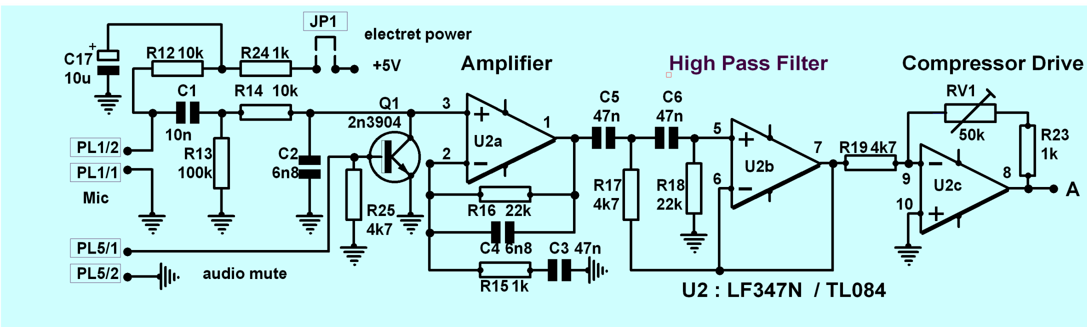

Figure 2.3 low level audio amplification and equalisation

Refer to figure 2.3 above. This section of the processor PCB consists of three op amp stages. U2a amplifies the input signal by about 22 times, driving U2b, a high-pass filter with zero gain.U2c provides further amplification sufficient to drive the compression circuit described in the next section. This stage provides a gain variation from +20dB to -10dB to cater for different mic sensitivities and compression requirements. U2a and U2b circuitry is inspired by Jame's Tonne's processing paper mentioned earlier, although the R/C values used differ.

C1/R13 is a passive high-pass filter, R14/C2 is a passive low-pass filter. This gives the combined response at U2 pin3 shown in figure 2.4 below. (modelled in LTSpice–IV)

U2a, as well as amplifying the signal, also tailors the frequency rsponse. The output from pin 1 looks like the blue trace below. The waveform is further modified with a sharper LF cut off from 300Hz downwards in U2b, finally producing the red waveform in figure 2.4.

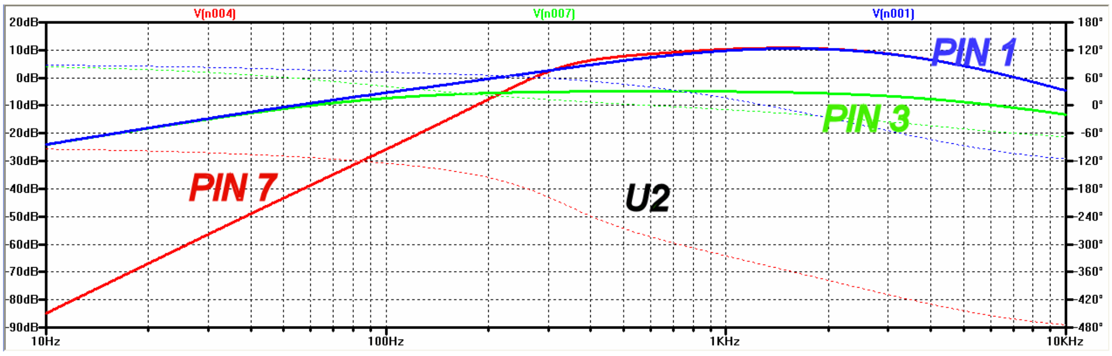

Figure 2.4: LT-Spice response curves of mic amp

JP1 applies electret power when needed. Q1 is a simple mic mute which can be integrated into a rx/tx sequencer, or used as a ‘privacy/cough’ mute.

2.2 Audio compressor

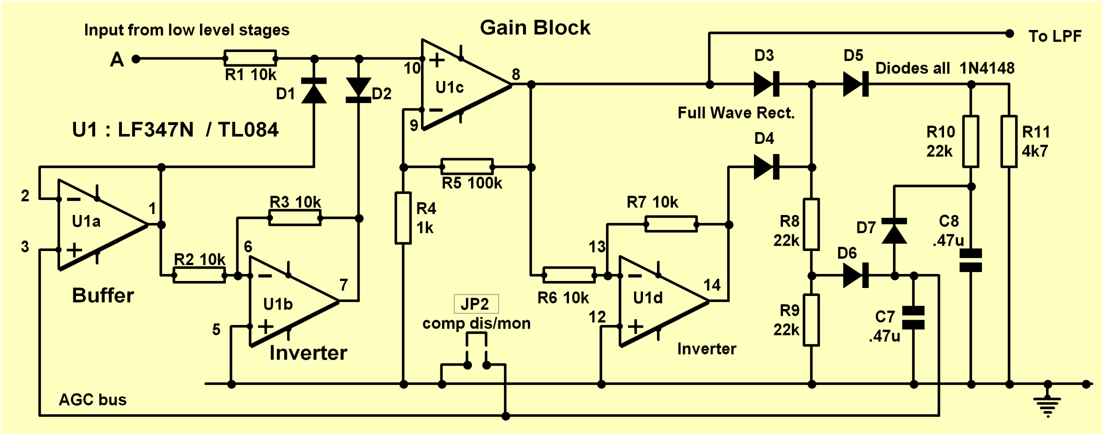

Figure 2.5 Compressor circuit

This part of the circuit is entirely the work of James Tonne, W4ENE. It is an RMS based AGC compression system with optimised attack, hold and decay time constants. James has chosen these well, and no pumping is apparent with sensible levels of compression.

Audio from the driver stage (at A) is fed to U1c which has a voltage gain of 100 times or 40dB (defined by R4/R5). However the level of audio at U1c pin10 can be controlled by the variable attenuator formed by R1 and D1/D2. The amount of attenuation is determined by how much forward bias is applied to the diodes. The diodes are in turn driven by the buffer U1a and inverting buffer U12b which are connected to the AGC bus. The AGC bus voltage varies with speech amplitude, forming a control loop. The relationship between the incoming audio and the varying AGC bus voltage defines the compression characteristic.

The AGC bus voltage is derived as follows: The output of U1c feeds D3 and (via the inverter U1d) D4, forming a full-wave rectifier so that both peaks and troughs are captured. The output from D3/D4 will vary dynamically with audio level, but to explain the circuitry, consider a steady DC voltage of say 4V. R8 / R9 will halve this voltage as it is applied to D6. C7 will charge via R8 and D6 to a level close to 2V. C7 is across the AGC bus and the charging time-constant of C7 is the compression attack-time.

The 4V in our example also charges C8 via D5 and R10. Ignoring any diode drop in D5, C8 will charge to the full 4V, back biasing D7. So, we now have 2V on the AGC bus and D7 reverse biased by 2V. Let us now remove the 4V. C7 cannot discharge via D6. The only path is via D7, but this is reverse biased. It is only reverse biased however until C8 discharges, via R10 and R11, to a point where D7 can start to conduct. This is the recovery delay. C7 can then discharge via D7, R10 and R11. This is the recovery time.

Simpler attack/decay circuits produce LF distortion because the AGC line is modulated by low frequency ripple. The recovery delay prevents this happening. With the component values used the attack time is 3ms, recovery delay 5ms and recovery time 10ms.

JP2 disables the AGC bus. This allows a higher audio level to be passed through the circuitry for modulator line-up. A constant tone has a high RMS value so the AGC bus settles to a level that gives a lower output than normal speech, making it impossible to fully modulate a transmitter on a whistle or single tone despite a high modulation level being produced on normal speech. Disabling the AGC bus also allows the gain stage to be driven into clipping should adventurous users wish to experiment with a more brutal approach to loudness enhancement (luckily U3 prevents too much spectrum carnage!).

The AGC bus voltage can be monitored from JP2; useful for compression level monitoring. The voltage could be applied to say an LED bargraph driver chip to indicate compression level. Note any circuitry added at this point must have a high impedance input so as not to load the AGC bus.

2.3 Low-pass filter

Audio emerging from the compressor has undergone amplitude compression and frequency response tailoring, but so far there is no 'brick-wall' filtering to limit higher frequencies. Voice frequencies will still be present that would result in an unacceptably wide modulation envelope. U3 remedies this.

U3 is an 8-pole switched-capacitor filter that can be adjusted to define the overall bandwith of the audio. With eight poles the roll-off is very steep. A single capacitor C9 alters the internal clock frequency that defines the filter corner frequency. There is a 100:1 clock to corner frequency relationship. Reducing C9 raises the clock frequency thus raising the corner frequency. 68p gives a corner of about 4500Hz.

The spare op amp stage in U3 is not used as the output impedance is fairly high and unsuitable as a final output stage. U2d buffers the audio, with C10 acting as a clock filter to remove any chance of U3 clock leakage.

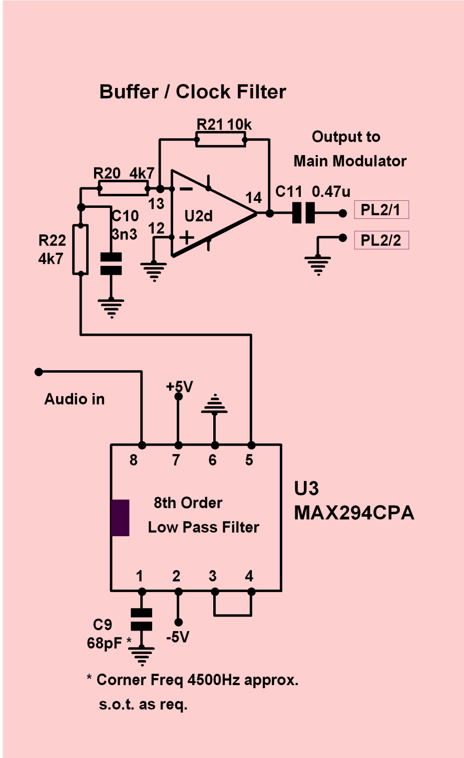

Figure 2.6 Low Pass Filter

2.4 Power supply

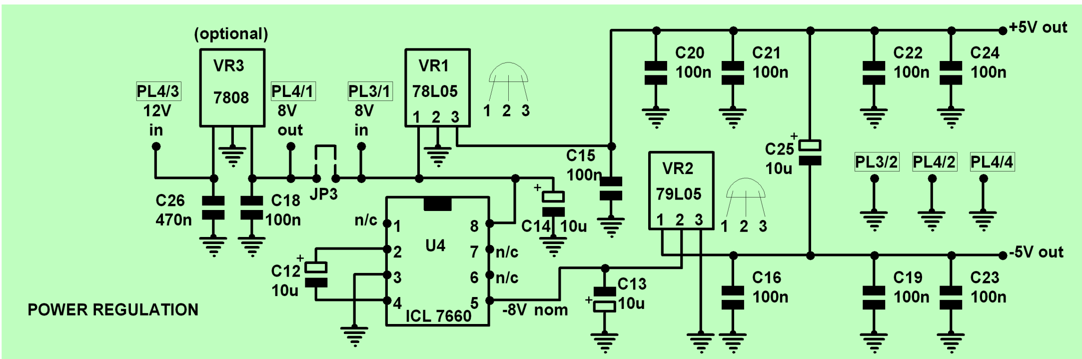

Figure 2.7: Power Supply

The final part of the speech processor to be described is the power supply section. The output requirement of the power supply is a +/- 5V rail for the audio ICs. The input can be either 8V supplied via PL3/1 from an external regulator, for instance one used to provide 8V for the FET drivers in the FAT5 RF PA or any voltage VR3 can stand (35V theoretically). If VR3 is used, a feed can be taken from PL4/1 to power the RF-PA drivers.

The +5V line is regulated by a standard low power linear regulator VR1. Similarly the -5V line is regulated by VR2. The voltage feed to VR2 is produced by U4 an industry standard DC-DC converter which produces slightly less than -8V on-load from a +8V input. VR2 stabilises the output at -5V with C13 acting as a reservoir capacitor. VR1 and VR2 ensure no switching transients from U4 appear on the 5V rails although extra decoupling may be needed at the main supply input if other sensitive circuitry is connected in parallel.
NB: The maximum supply voltage for U4 is 12V so take care if an external 8V is used via PL3/1.

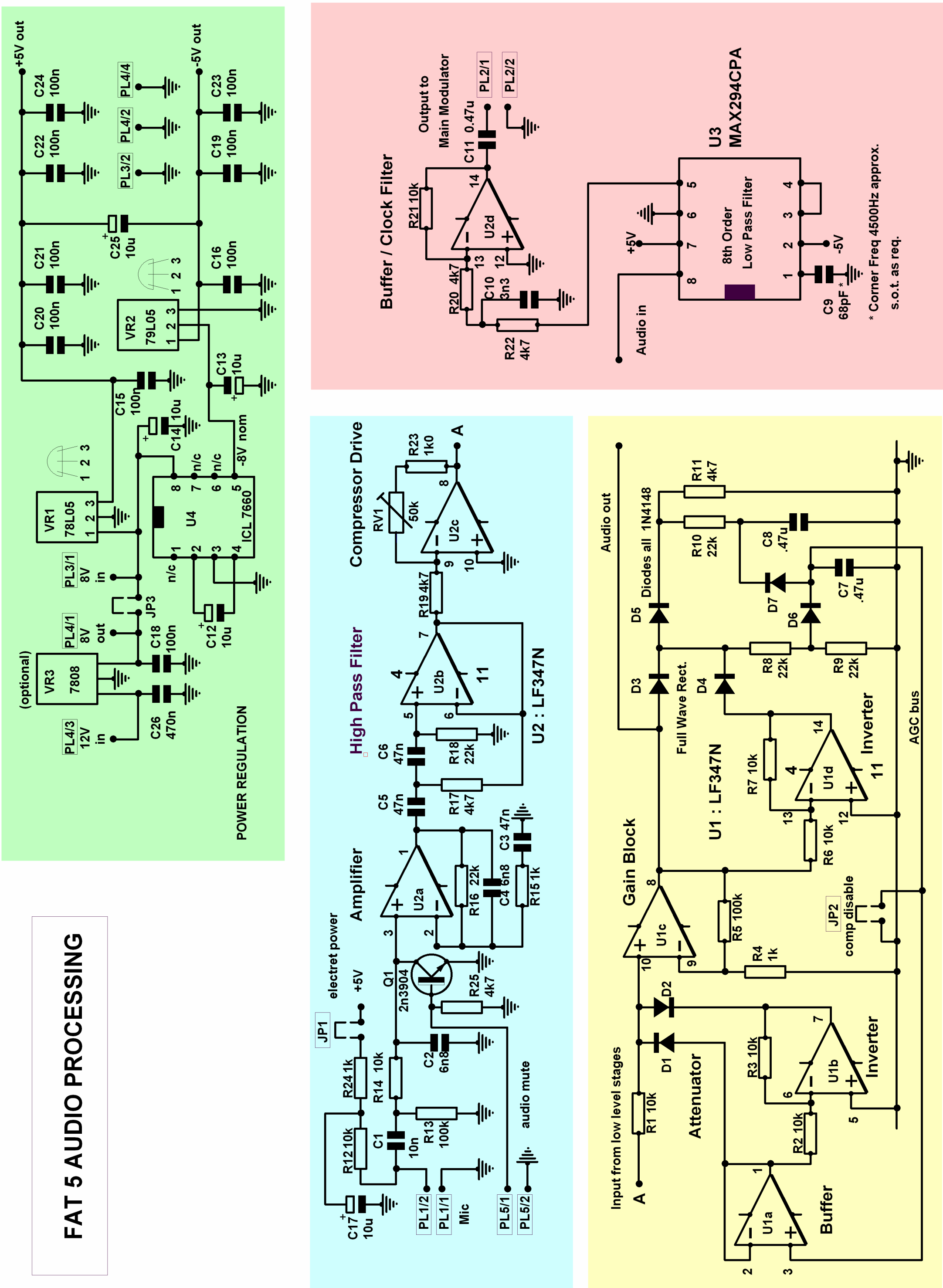

Figure 2.8 Speech Processor complete circuit diagram

2.5 PCB Layout

Print this out to assist with component placement

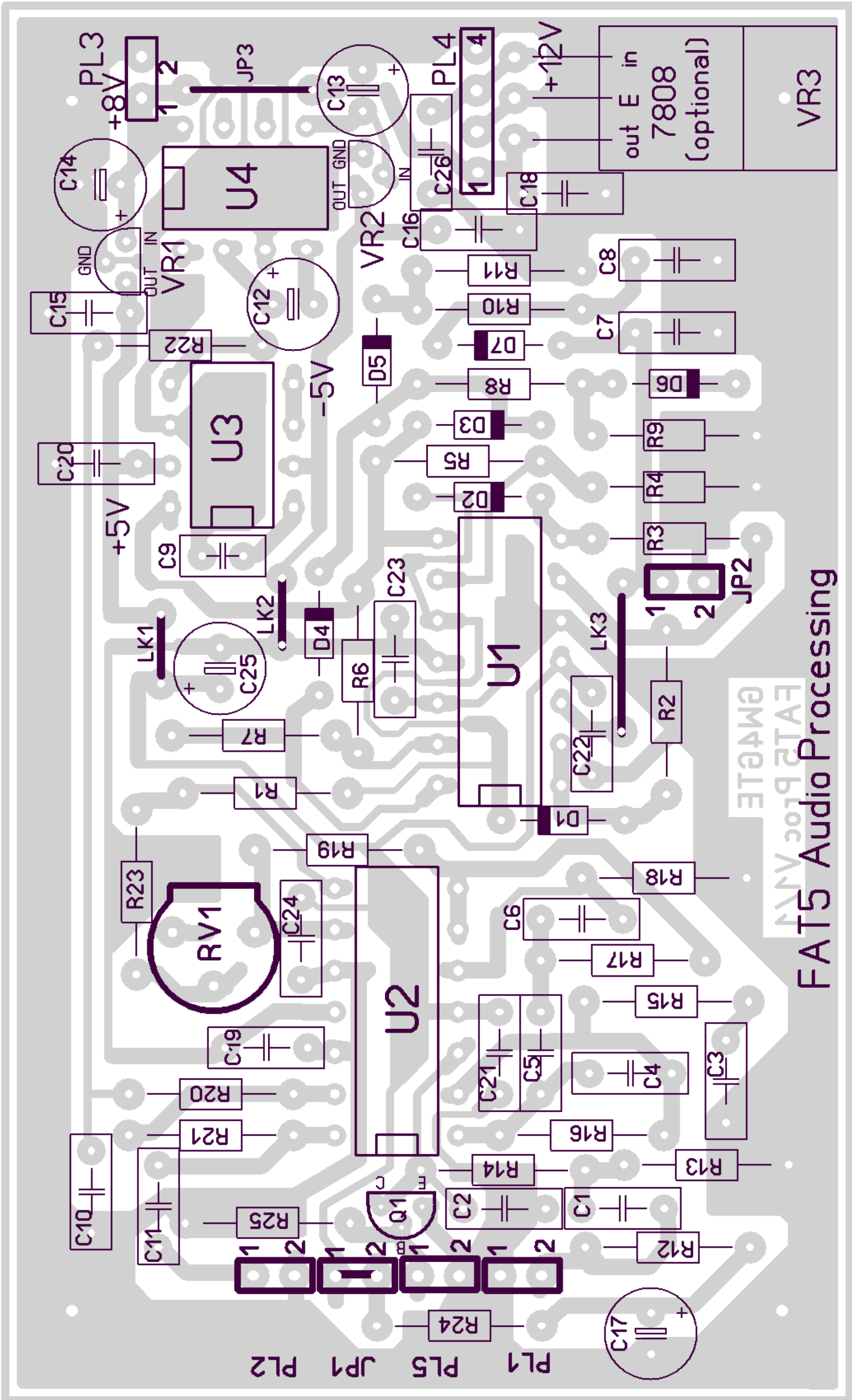

Figure 2.9 Speech Processor PCB layout
[Note the three wire links LK1,2,3]

2.6 Processor PCB pin connections

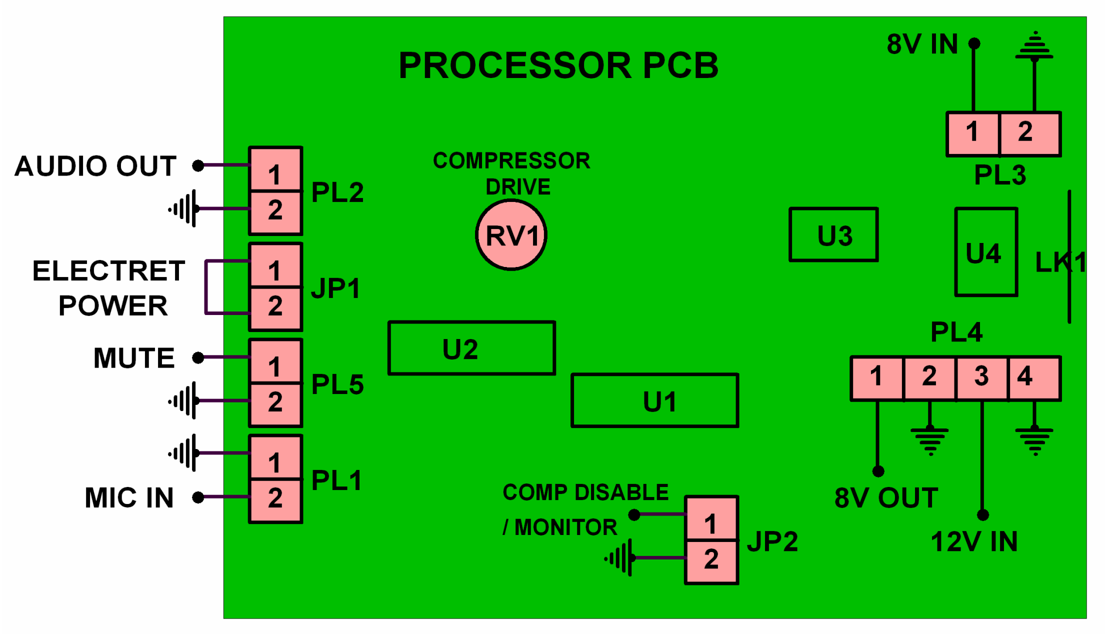

Figure 2.10 PCB connections

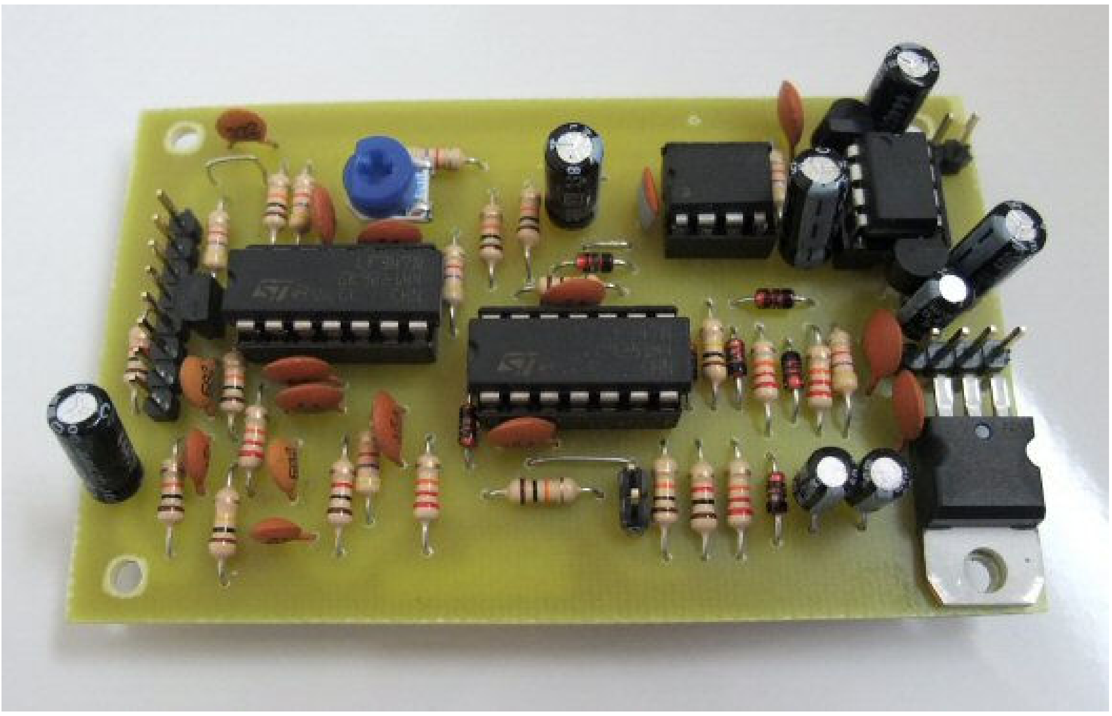

Figure 2.11 Fully populated processor PCB

(Ignore the fact that C11, top left, is a link in this version. Actual components supplied may differ physically.)

Connector	Use	Notes
PL1 / 1	0V	
PL1 / 2	Mic input	Make JP1 for electrets
PL2 / 1	Audio out	
PL2 / 2	0V	
PL3 / 1	8V in	Disable LK1 if used
PL3 / 2	0V	
PL4 / 1	Regulated 8V out	To power PA driver chips
PL4 / 2	0V	
PL4 / 3	12V in	Min 10V. recc max 15V
PL4 / 4	0V	
PL5 / 1	Audio mute	Apply 5V via 3k3 resistor
PL5 / 2	0V	
JP1 / 1	Electret power from 5V	Connect to JP1 / 2
JP1 / 2	Electret power to R24	Use handbag link provided
JP2 / 1	Comp disable / monitor	Connect to JP2 / 2 to disable
JP2 / 2	0V	
JP3	Wire link when VR3 is used	Connects 8V from VR3
LK1,2,3	bridges between track sections	Make these during construction

Table 2.1 Connector functionality

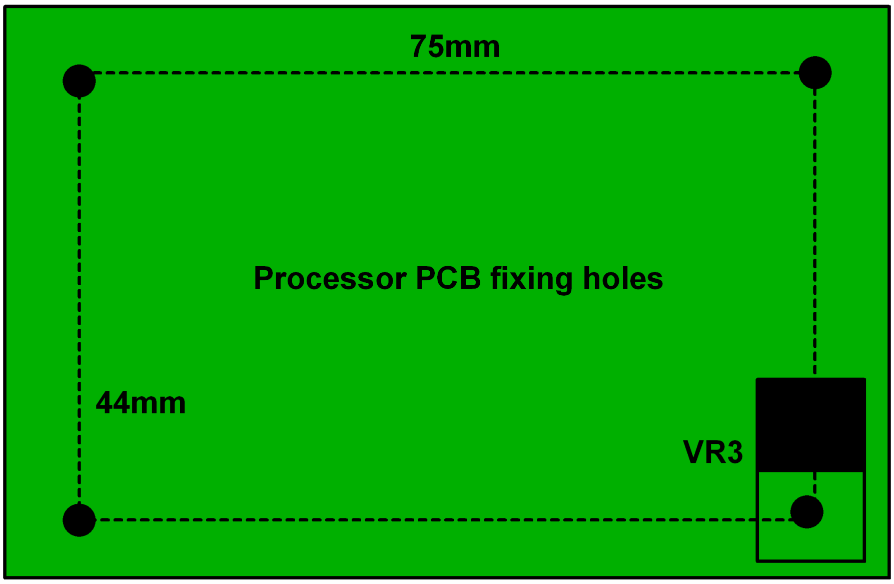

Figure 2.12 FAT-MAX hole drilling dimensions (not to scale)

2.7 Component List

Print this out to assist with component identification

Component	Value	Marked as
R1	10k	
R2	10k	
R3	10k	
R4	1k0	
R5	100k	
R6	10k	
R7	10k	
R8	22k	
R9	22k	
R10	22k	
R11	4k7	
R12	10k	
R13	100k	
R14	10k	
R15	1k0	
R16	22k	
R17	4k7	
R18	22k	
R19	4k7	
R20	4k7	
R21	10k	
R22	4k7	
R23	1k0	
R24	1K0	
R25	4k7	
RV1	50K	
C1	10n ceramic 5mm pitch	
C2	6n8 ceramic 5mm pitch	
C3	47n ceramic 5mm pitch	
C4	6n8 ceramic 5mm pitch	
C5	47n ceramic 5mm pitch	
C6	47n ceramic 5mm pitch	
C7	470n ceramic 5mm pitch	
C8	470n ceramic 5mm pitch	
C9	68p ceramic 2.54mm pitch	
C10	3n3 ceramic 5mm pitch	
C11	470n ceramic 5mm pitch	
C12	10uF 16V radial elect 2.54mm pitch	
C13	10uF 16V radial elect 2.54mm pitch	
C14	10 uF 16V radial elect 2.54mm pitch	
C15	100n ceramic 5mm pitch	
C16	100n ceramic 5mm pitch	
C17	10uF 16V radial elect 2.54mm pitch	
C18	100n ceramic 5mm pitch	Use with optional VR3
C19	100n ceramic 5mm pitch	
C20	100n ceramic 5mm pitch	
C21	100n ceramic 5mm pitch	
C22	100n ceramic 5mm pitch	
C23	100n ceramic 5mm pitch	
C24	100n ceramic 5mm pitch	
C25	10uF 16V radial 2.54mm pitch	
C26	470n ceramic 5mm pitch	

D1	1N4148	
D2	1N4148	
D3	1N4148	
D4	1N4148	
D5	1N4148	
D6	1N4148	
D7	1N4148	
U1	LF347N op amp (or TL084)	
U2	LF347N op amp (or TL084)	
U3	MAX294CPA sw cap filter	
U4	ICL7660 DC-DC converter	
VR1	78L05 +5V reg TO92	
VR2	79L05 -5V reg TO92	
VR3	7808 (optional)	
Q1	2N3904 or similar GP NPN	Check pinout if alternative used

Table 2.2 Speech Processor Component List

2.8 PCB Links

2.8.1 Electret Power JP1

The microphone input can cope with dynamic microphones and electret microphones. When an electret microphone is used make the link JP1 to power the electret. A handbag link should have been provided in the kit.

2.8.2 Compressor disable/monitor JP2

JP2 when made, disables the compressor. This is required to set up the main modulator and is also used during initial testing of the audio processor. See section 2.5 below for details on modulator adjustment.

JP2 is also a DC level monitoring point for the compressor function. This is used for compression setup. An LED bargraph/driver IC could be used to display comp level.

2.8.3 Regulated 8V rail JP3

JP3 is a wire link that connects the regulated 8V from VR3 to the main supply input at PL3/1. Make this connection if you are using VR3 to supply power to the rest of the PCB. If you intend to use VR3 to supply 8V to the FET drivers in the FAT5 RF-PA you will need to add a small heatsink or mount the regulator on a separate heatsink away from the main PCB and backfeed the processor PCB via PL3/1.

2.8.4 Microphone Muting PL5/1

Q1 acts as a mic mute. Apply a voltage (say 5V) via a series resistor of 3k3 (value not critical) to PL5/1 to mute the microphone. It is desirable to mute the mic when on standby.

NOTE: Do not apply 5V directly to PL5/1 or the base/emitter junction of Q1 will be destroyed. Use the recommended 3k3 series resistor in series.

2.9 Processor Testing

Visual

(__) After completing construction carefully inspect the PCB for any missed joints or solder bridges - particularly where thinner track passes between two solder pads.

Check for correct voltages and current

(__) Before inserting U1,U2 and U3, apply power in the manner decided upon and check the correct voltages are being generated by the regulator circuitry. Then plug in U1,U2 and U3 and check the supply current is around 30mA.

Check Electret power

(__) Temporarily make link JP1. Ensure 5V is present on PL1/2 then remove the link.

Check op amp offset voltage

(__) Check the voltage on U1 pin 8 against earth. With RV1 set to minimum the offset should be less than 1V. (Any offset voltage on U1 pin 10 is magnified by the gain of U1c)

2.9.1 Functional Test

There is no alignment required, so you don't need to carry out these tests, but you will better understand the processor if you do. At least bear the tests in mind if you subsequently find something is amiss.

Equipment required:

- Oscilloscope, Y trace at 1V/div, timebase to 200us
- Audio Signal Generator initially set to 1kHz, zero output

Preparation

- Set RV1 to mid-travel and ensure JP1 and JP2 are open circuit.
- Connect the oscilloscope to PL2/1 and PL2/2 (earth).
- Connect the audio generator to PL1/2 and PL1/1(earth).

Compressor function

(__) Increase the generator audio level (only a few millivolts will be required) until an output is seen on the oscilloscope trace. Ensure it is a pure sine wave.

(__) Continue to advance the audio level until the output level stops increasing. Note the output voltage level, which should be around 3V Pk-Pk.

(__) Increase the audio level by 20dB (i.e. ten times the previous level). Ensure the output remains undistorted and the level only increases by around 10%. Reduce the signal generator level to zero.

Output level

(__) Apply a link to JP2 to disable the compressor circuit.

(__) Increase the audio level until clipping occurs. Note the peak-to-peak output level at which clipping just starts to occur (depending on any voltage offset present, one side of the waveform will clip first) A level of around 6V Pk-Pk should be achieved before clipping onset.

(__) With JP2 still connected, turn RV1 fully clockwise. reducing the audio input so that the waveform is not clipped. Note the level. Now turn RV1 fully counter-clockwise. The output level should decrease by 15 times (-34dB).

(__) Return RV1 to mid position and adjust the audio input to produce an output of around 1V Pk-Pk

8-pole filter corner frequency

(__) Watching the oscilloscope trace, sweep the audio input from 3kHz to 5kHz. Depending on the exact value of C9, the output will abruptly drop at a particular frequency. This should be around 4500Hz with C9 at 68pF. C9 can be adjusted for any transmit bandwidth desired.

Frequency response

(__) Now sweep the audio from 10Hz to the filter knee established in the previous test. Note the lower frequencies are attenuated, a peak is reached at about 1500Hz then a more gentle roll-off occurs towards the filter knee The difference in output level between 100Hz and 1kHz shoud be 35dB (i.e. 1kHz is 10 times higher in voltage level).

Audio-mute check

(__) **NB: DO NOT OMIT THE 3K3 RESISTOR DURING THIS TEST.** Set the audio generator to 1kHz, to give a 1V P-P output. Apply 5V to PL5/1 via a 3k3 resistor and note the output falls to a very low level.

(__) Remove link JP2

2.10 Setting the compression level.

The aim is to find the right setting where talk power is substantially increased without it becoming obvious a speech processor is in use. Excessive compression merely increases background noise during breaks in speech and adds nothing to readability.

There is only one potentiometer (RV1) to adjust on the processor board. RV1 controls the gain of the compressor driver amp. The gain of this stage can be adjusted over a 30dB range from +20dB to -10dB.

An effective (albeit unscientific) adjustment method is as follows:
Using an oscilloscope or AC voltmeter to monitor the output amplitude of the processor, speak normally at your usual distance from the microphone and note the level. Adjust RV1 so that the level is maintained when you double your distance from the microphone, just starting to fall as you back further away. Doubling the distance should decrease the sound energy level by 6dB, so if the compressor AGC can compensate for the drop the compression can be said to be 6dB. Don’t whistle into the mic. A sustained ‘ahhhhhhhhhh’ is fine if it’s at your normal voice level.

From here on you may wish to obtain a few on-air reports and make adjustments of RV1. Don’t rely on just one report though, and favour reports from stations who can measure, not guess.This assumes you have set up your transmitter modulation level correctly – see the sections below covering the two modulator PCBs. With a complete transmitter the test may be easier to carry out using a modulation meter (using a dummy load of course!)

JP2 can be used to monitor the compression level. Use an oscilloscope set to DC input, timebase set to 1ms, Y input to 100mV/div and X-trigger disabled or set to another channel. Make sure a high impedance device is attached to JP2 otherwise the AGC bus will not function correctly. If your probe has X1 and X10, use X10. Note how the level fluctuates as you speak. The change of level is more logarithmic than linear. An LED bargraph circuit could be added to display this level.

2.11 Component adjustments.

It's always possible a combination of microphone and the vagaries of the human voice conspire to make the end result not quite right, in which case there are simple component changes you can try. The changes have been chosen such that components can be soldered in parallel with the existing parts. There is no need to remove any components.

Compression level
In the unlikely event RV1 provides too little or too much gain at any setting, make the following component alterations:

> **Too much gain.**
> If there is too much compression with RV1 at minimum, decrease the value of R23. Try bridging it with say 470ohms rather than attempting to remove the component.
>
> **Too little gain.**
> If there is not enough compression even with RV1 at maximum, decrease the value of R19. Try bridging it with another 4k7 resistor first of all. (increasing R23 will have the same effect but you will need to unsolder the original part and R19 offers a greater range of adjustment)

Audio quality 'too thin/toppy'
Try increasing C5 and C6 to 100n. Do both together. This will alter the LF knee from 300Hz to 100Hz

Audio quality 'too bassy'
This is very unlikely unless your microphone is very bassy. Try backing away from the mic a little (bass frequencies fall away more quickly as the longer wavelengths are less directional)

Or, just try another microphone …!

Builders who are familiar with LTSpice IV or equivalent programs may wish to model the low-level stages of the processor and 'play about' with component values. Please contact the author for a Spice file of the circuit. Figure 2.4 shows a typical response plot from LTSpice IV.

3. LOW / MEDIUM POWER MODULATOR

The TDA2050 modulator is a good solution for low/medium power transmitters. The limitation for higher power is the maximum TDA2050 operating voltage of 50V. When used in conjunction with the FAT5 RF-PA module with all FETs installed it is unlikely this voltage will be exceeded. Unlike the LM3886, the TDA2050 does not have internal mute circuitry for low-voltage cut-out thus allowing a QRP transmitter to run off say 13.8V (10W carrier can still be expected at this voltage however)

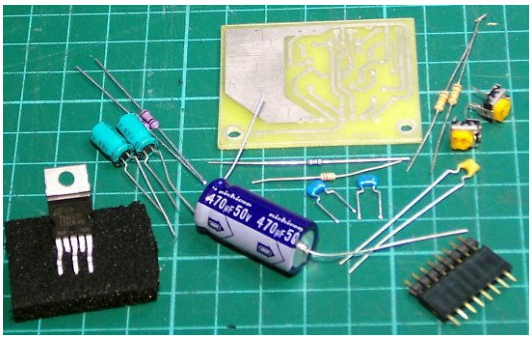

Figure 3.1 TDA2050 Modulator kit of parts

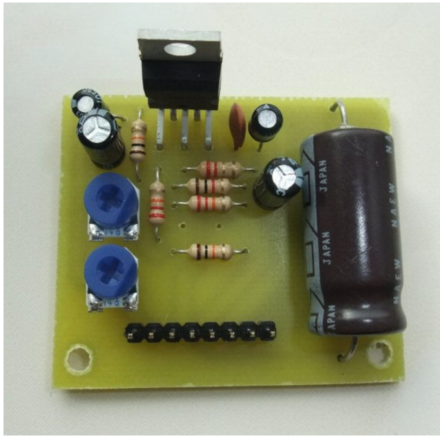

Figure 3.2 Assembled TDA2050 Modulator

3.1 Circuit Description / diagram

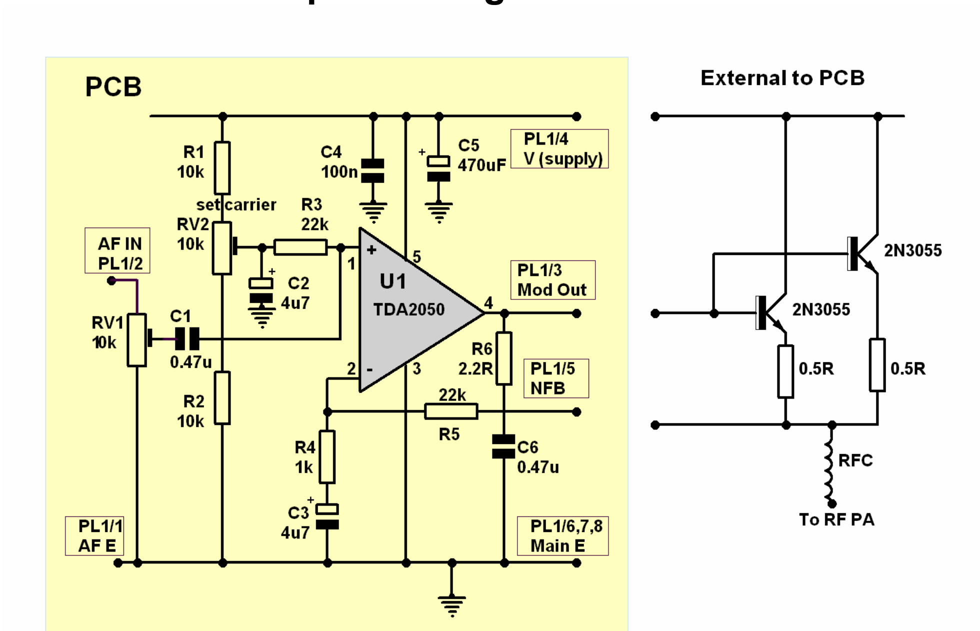

Figure 3.3 TDA2050 circuit Diagram

The circuit is essentially a single IC power op-amp with supporting circuitry, biased to operate from a single rail. The output (pin 4) is set for a standing voltage of around half-rail by means of RV2. C3 isolates R4 making the gain at DC unity. Under AC conditions R5 and R4 define the gain as 22. R6 and C6 are added as per the device data sheet to form a pole to prevent high frequency oscillations. The amplifier is non-inverting as the audio input is fed to the positive input of the IC. See Section 5 below for interfacing.

3.2 Construction Notes

Before commencing construction
Visually check the PCB for any etching or drilling errors. While all boards are carefully checked, as they are all made by hand a double-check is sensible. Any PCBs found defective will be replaced free of charge

There is no specific assembly order for the PCB, however following the component list and ticking off parts as they are installed is suggested.

The modulator PCB has been tested and found to be stable, however this type of amplifier can become unstable if care is not given to earth routing when part of a complete transmitter. Specifically, try to keep input currents away from output currents. The data sheets on these chips gives good advice.

The heatsink tab on the IC should be either isolated or connected to 0V. This means an isolating washer is not required. Please double check before attaching the module to a grounded structure. Also bear this in mind when considering earth currents.

3.3 Modulator Testing and setup

(__) After completing construction carefully inspect the PCB for any missed joints or solder bridges - particularly where thinner track passes between two solder pads.

(__) Using the circuit diagram and layout diagram carry out ohmic tests for short circuits.

3.3.1 Stand-alone functionality check

To test the PCB as a stand-alone module you will need the following equipment:

- Oscilloscope
- Audio signal generator
- Multi-meter
- Power supply (A standard 13V8 supply will suffice for the moment)

Preparation

(__) Turn RV1 and RV2 to zero (fully counter-clockwise).

(__) Connect PL1/5 to PL1/3 to complete the negative feedback loop.

(__) Connect the multi-meter to PL1/3; use a DC voltage range to match the psu voltage.

Carrier voltage adjustment – test and preset

(__) Connect the power supply and switch on. The meter on PL1/3 should read approximately one third of the supply voltage.

(__) Turn RV2 fully clock-wise. The meter on PL1/3 should read approximately two thirds of the supply voltage.

(__) Subtract two volts from the main supply voltage then divide by two. e.g. 13.8v – 2V = 11.8V. Divided by two = 5.9V. Adjust RV2 so that the meter on PL1/3 reads the calculated figure.

Set modulation symmetry

(__) Connect the audio signal generator to PL1/2, set to 1kHz and zero level.

(__) Set RV1 to mid-position.

(__) Disconnect the meter from PL1/3 and attach the oscilloscope instead.

(__) Increase the audio input level and observe a sine wave on the oscilloscope.

(__) Continue to increase the audio level until clipping is observed on the trace.

(__) Adjust RV2 so that clipping occurs simultaneously on peaks and troughs.

(__) **Note this method of setting up.** This is also used when the PCB is driving the external power transistors. The correct adjustment point of RV2 will change when a load is connected.

3.4 PCB Layout

Print this out to assist with component placement

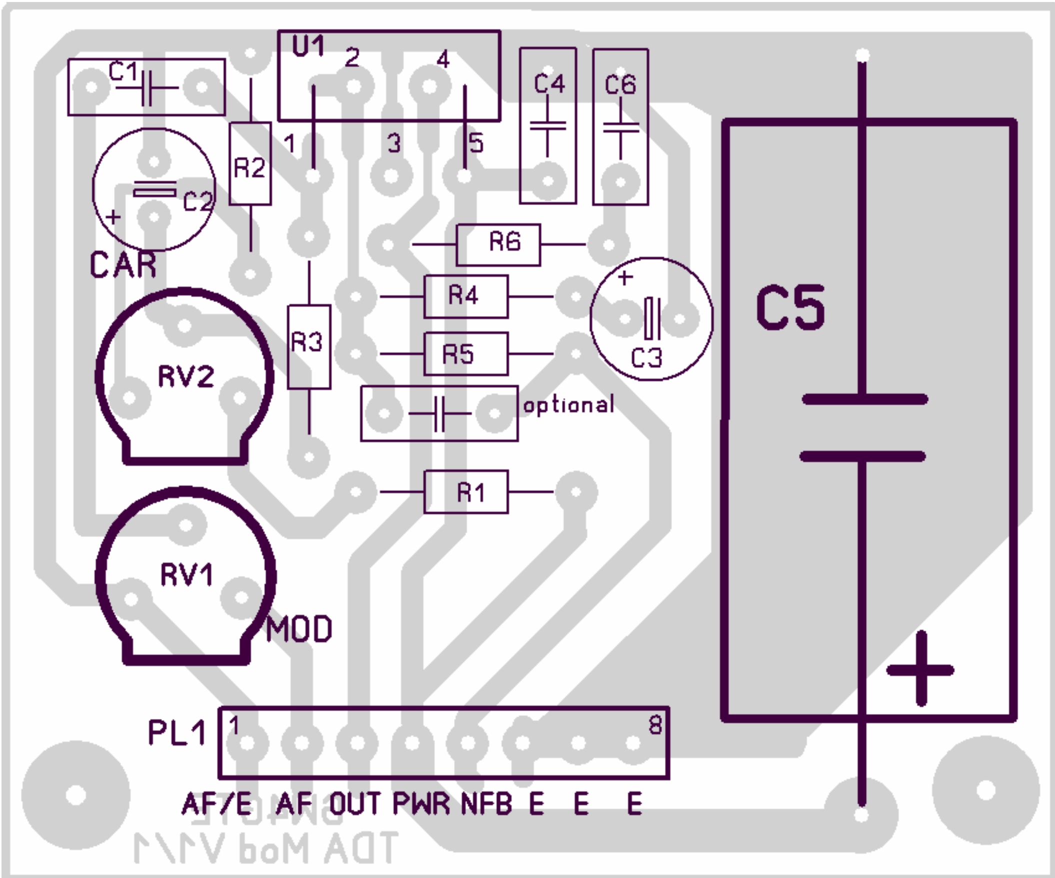

Figure 3.4 Component Layout (TDA2050)

3.5 Component List (PCB)

Print this out to assist with component identification

Component	Value
R1	10k ¼ W carbon film, 5%
R2	10k ¼ W carbon film, 5%
R3	22k ¼ W carbon film, 5%
R4	1k ¼ W carbon film, 5%
R5	22k ¼ W carbon film, 5%
R6	2R2 ¼ W carbon film, 5%
RV1	10K lin 5mm pitch
RV2	10k lin 5mm pitch
C1	470n ceramic 5mm pitch
C2	4u7 radial electrolytic 2.5mm pitch
C3	4u7 radial electrolytic 2.5mm pitch
C4	100n ceramic 5mm pitch
C5	470uF axial electrolytic 30mm x 15mm max
C6	470n ceramic 5mm pitch
U1	TDA2050V
PL1	8 pin 2.54mm pitch header strip

Table 3.1 Component List (TDA2050)

4. MEDIUM / HIGH POWER MODULATOR

This is the alternative modulator PCB to the TDA2050 version. Although slightly more complex, the configuration is essentially the same. The LM2886 (or LM2876/3876) IC will provide greater driving power than the TDA2050 and will stand a substantially higher supply voltage. The downside is there is a minimum suppy voltage of around 24V below which the IC's internal low voltage cut-out kicks in. Higher power designs using higher voltage FETs in the RF PA should use this version. This amplifier is however suitable for the standard FAT5 RF-PA configuration delivering 50W carrier from a PA voltage of around 14V as the main supply rail will be around 32V, significantly greater than the 24V minimum.

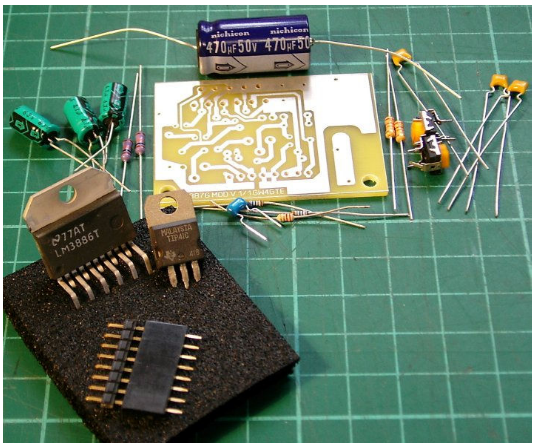

Figure 4.1 L3886 Modulator kit of parts

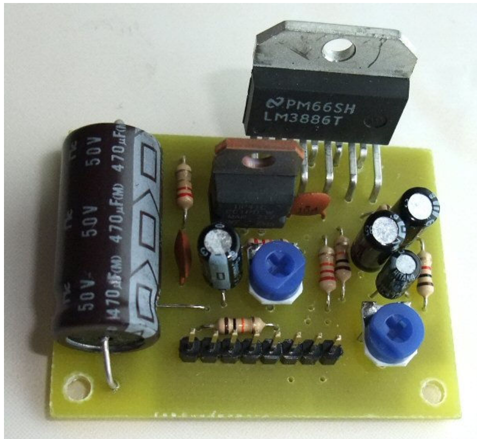

Figure 4.2 Assembled LM3886 modulator

4.1 Circuit Description / diagram

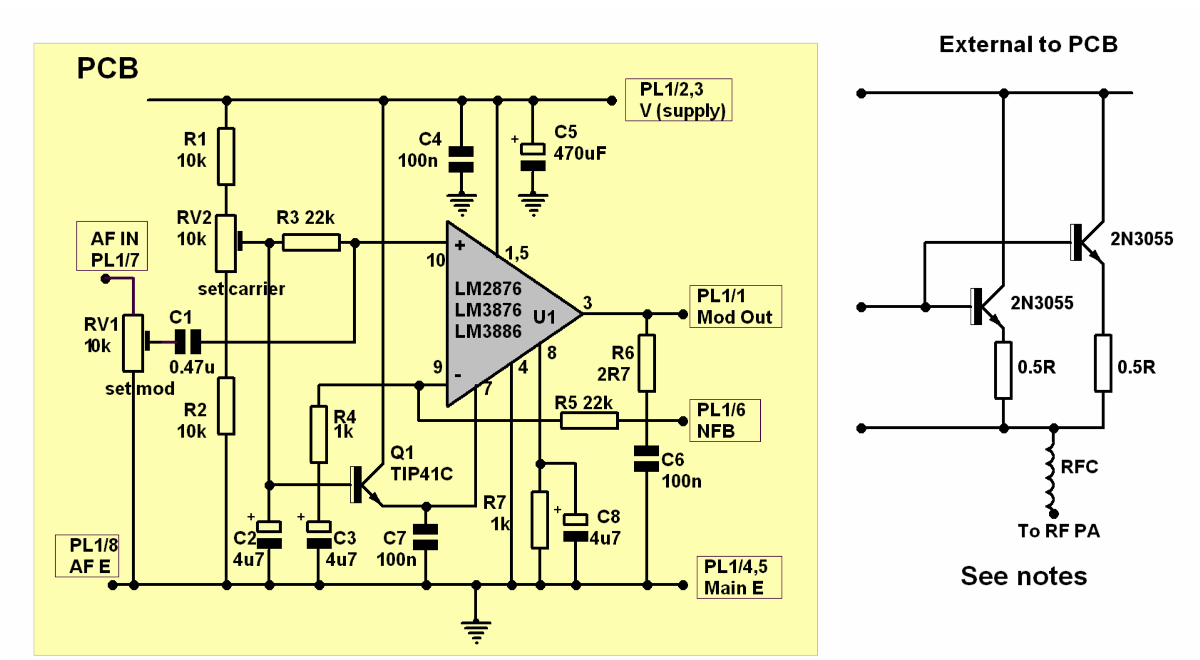

Figure 4.3 Circuit Diagram (LM3886)

The circuit is essentially a single IC power op-amp with supporting circuitry, biased to operate from a single rail. The output (pin 3) is set for a standing voltage of around half-rail by means of RV2. C3 isolates R4 making the gain at DC unity. Under AC conditions R5 and R4 define the gain as 22. R6 and C6 are added as per the device data sheet to form a pole to prevent high frequency oscillations. The amplifier is non-inverting as the audio input is fed to the positive input of the IC. Q1 is a common emitter current amplifier to establish a low impedance virtual earth at pin 7.In most other respects the modulator closely resembles the TDA2050 circuit. See Section 5 below for interfacing.

4.2 Construction Notes

Before commencing construction
Visually check the PCB for any etching or drilling errors. While all boards are carefully checked, as they are all made by hand a double-check is sensible. Any PCBs found defective will be replaced free of charge

Apart from soldering LK1 in position before attaching C5, there is no specific assembly order for the PCB, however following the component list and ticking off parts as they are installed is suggested.

The modulator PCB has been tested and found to be stable, however this type of amplifier can become unstable if care is not given to earth routing when part of a complete transmitter. Specifically, try to keep input currents away from output currents. The data sheets on these chips gives good advice.

The heatsink tab on the IC should be either isolated or connected to 0V. This means an isolating washer is not required. Please double check before attaching the module to a grounded structure. Also bear this in mind when considering earth currents.

4.3 Modulator Testing and setup

(__) After completing construction carefully inspect the PCB for any missed joints or solder bridges - particularly where thinner track passes between two solder pads.

(__) Using the circuit diagram and layout diagram carry out ohmic tests for short circuits.

4.3.1 Stand-alone functionality check

To test the PCB as a stand-alone module you will need the following equipment:

- Oscilloscope
- Audio signal generator
- Multi-meter
- Power supply (A minimum of 24V is required)

Preparation

(__) Turn RV1 and RV2 to zero (fully counter-clockwise).

(__) Connect PL1/1 to PL1/6 to complete the negative feedback loop.

(__) Connect the multi-meter to PL1/1; use a DC voltage range to match the psu voltage.

Carrier voltage adjustment – test and preset

(__) Connect the power supply (set to at least 24V) and switch on. The meter on PL1/1 should read approximately one third of the supply voltage.

(__) Reduce the power supply voltage. At a little under 24V the meter reading should drop to almost zero. Restore the supply voltage. This tests the IC's low voltage mute function. (this 'feature' prevents the IC being used at lower voltage levels)

(__) Turn RV2 fully clock-wise. The meter on PL1/1 should read approximately two thirds of the supply voltage.

(__) Take four volts from the main supply voltage then divide by two. e.g. 24V – 4V = 20V. Divided by two = 10V. Adjust RV2 so that the meter on PL1/1 reads the calculated figure.

Set modulation symmetry

(__) Connect the audio signal generator to PL1/7, set to 1kHz and zero level.

(__) Set RV1 to mid-position.

(__) Disconnect the meter from PL1/1 and attach the oscilloscope instead.

(__) Increase the audio input level and observe a sine wave on the oscilloscope.

(__) Continue to increase the audio level until clipping is observed on the trace.

(__) Adjust RV2 so that clipping occurs simultaneously on peaks and troughs.

(__) **Note this method of setting up.** This is also used when the PCB is driving the external power transistors. The correct adjustment point of RV2 will change when a load is connected.

4.4 PCB Layout

Print this out to assist with component placement

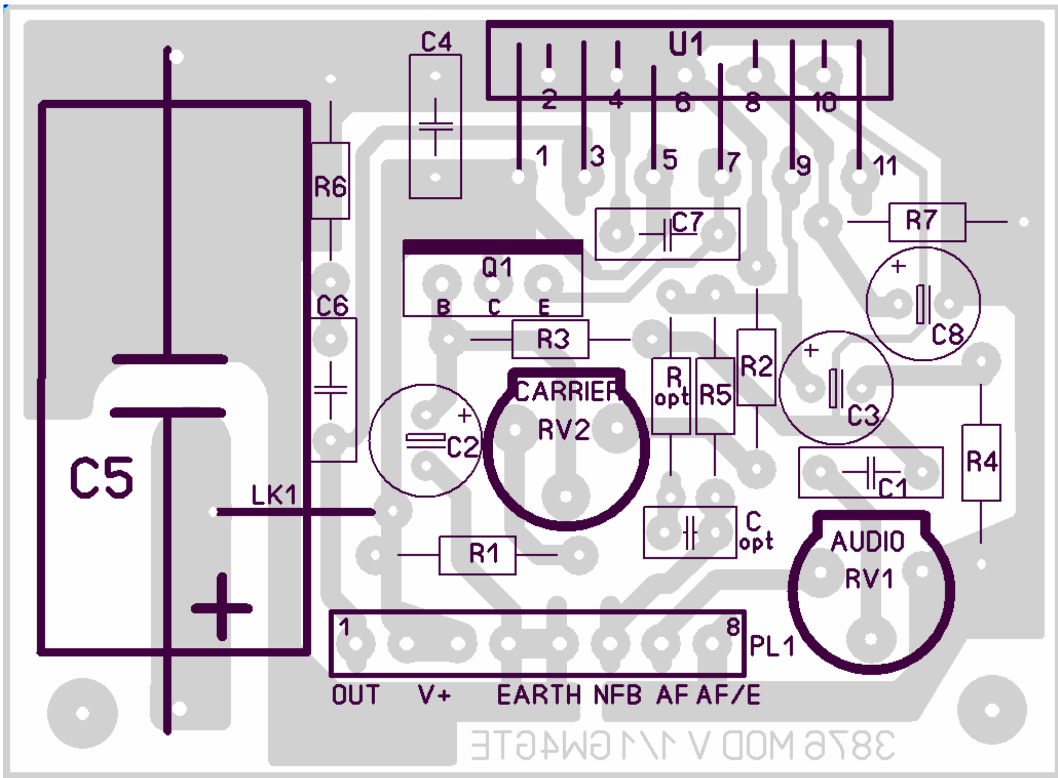

Figure 4.4 Component Layout (LM3886)

4.5 Component List (PCB)

Print this out to assist with component identification

Component	Value
R1	10k ¼ W carbon film, 5%
R2	10k ¼ W carbon film, 5%
R3	22k ¼ W carbon film, 5%
R4	1k ¼ W carbon film, 5%
R5	22k ¼ W carbon film, 5%
R6	2R7 ¼ W carbon film, 5%
RV1	10K lin 5mm pitch
RV2	10k lin 5mm pitch
C1	470n ceramic 5mm pitch
C2	4u7 radial electrolytic 2.5mm pitch
C3	4u7 radial electrolytic 2.5mm pitch
C4	100n ceramic 5mm pitch
C5	470uF axial electrolytic 30mm x 15mm max
C6	100n ceramic 5mm pitch
C7	100n ceramic 5mm pitch
C8	4u7 radial electrolytic 2.5mm pitch
U1	LM3876T/ LM3886T
Q1	TIP41A
PL1	8 pin 2.54mm pitch header strip

Table 4.1 Component list

5. MODULATOR INTERFACING

5.1 Power Transistors

The modulator PCB (of either type) does not supply current directly to the RF PA stage. PA current is handled by one or more series-pass power transistors whose bases are driven by the modulator PCB. The number of transistors required depends upone their type and the power level envisaged.

For a 10W carrier transmitter running off 13.8V a single transistor will be adequate. For 25W carrier two are recommended, and a minimum of three transistors should be used for a 50W carrier transmitter. Try to select transistors with similar gain characteristics if possible. (many multi-meters have basic transistor checking functions).

The type of transistor used is not critical as this is just audio. Any general purpose NPN power transistor that has a Vceo greater than the supply voltage should be fine. The TO3 case (traditional 2N3055 style) has a greater surface area than more modern packages and heat dissipation is better. An alternative is TIP3055 in a TO-247 package (a larger version of the TO-220 package the RF-PA FETs use). The TO-247 has the advantage of only needing a single hole fixing. Mica washer kits for TO-220 will usually fit the larger TO-247 as well. Mica heatsink washers are more efficient than the silicone rubber types.

5.2 Emitter Resistors

When more than one power transistor is used emitter resistors should be added to balance the current. The value used depends on the amount of gain mis-match present. A value of 0.47 ohms has worked fine in practice. The current differential between the transistors can easily be measured by comparing the voltage drop across each emitter resistor. Use a wirewound resistor with a suitable dissipation for the currents expected.

For a PA current of 5A and three transistors, each resistor is passing say 2A worse case (allowing for transistor imbalance). W = I x I x R so, W = 2 x 2 x 0.5 = 2W. A 3W min. resistor is recommended. One example is Rapid 62-0225 0.47R 3W W/W resistor @13p each.

The resistors can be mounted on a piece of Vero board or a tag strip. Space the resistors away from the board to assist cooling. Smaller resistors could be placed in the airstream of a heatsink cooling fan.

5.3 Heatsink

The power transistors must be mounted on a suitable heatsink The heat generated by the transistors is slightly more than the PA’s DC input as the transistors are in series with the PA and form the upper part of a potential divider with the RF PA volts sitting at the centre point.

For a 50W RF transmitter the heatsink will need to dissipate something in excess of 60W. If a large metal case is used to house the transmitter this may prove adequate although a greater thermal mass next to the transistors will be better for reducing thermal shock. Consider assisted cooling by means of a small DC fan; typical 12V computer case/psu fans are suitable. This dramatically improves heat removal. Blow air onto the heatsink rather than sucking air from it.

Heatsinks are regularly available from eBay at good prices. Alternatively a DIY heatsink can be made from strips of angle aluminium bolted together (e.g. B&Q stock 1m lengths)

How big? Two examples of a heatsink suitable for 50W RF (with fan cooling) are shown below.

The author's 100W transmitter uses a larger fan assisted heatsink with a thermistor attached drivng a fan-speed control circuit to reduce fan speed (thus noise) when not needed.

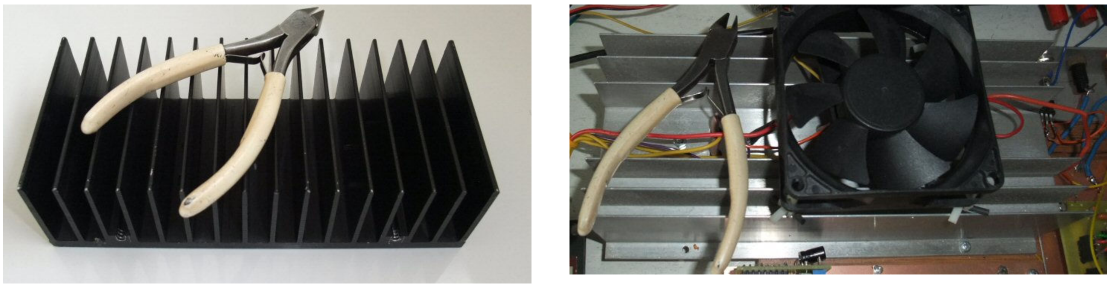

Figure 5.1 Example of heatsinks suitable for 50W TX (when fan assisted)

5.4 Integrating the modulator into a complete FAT5 system

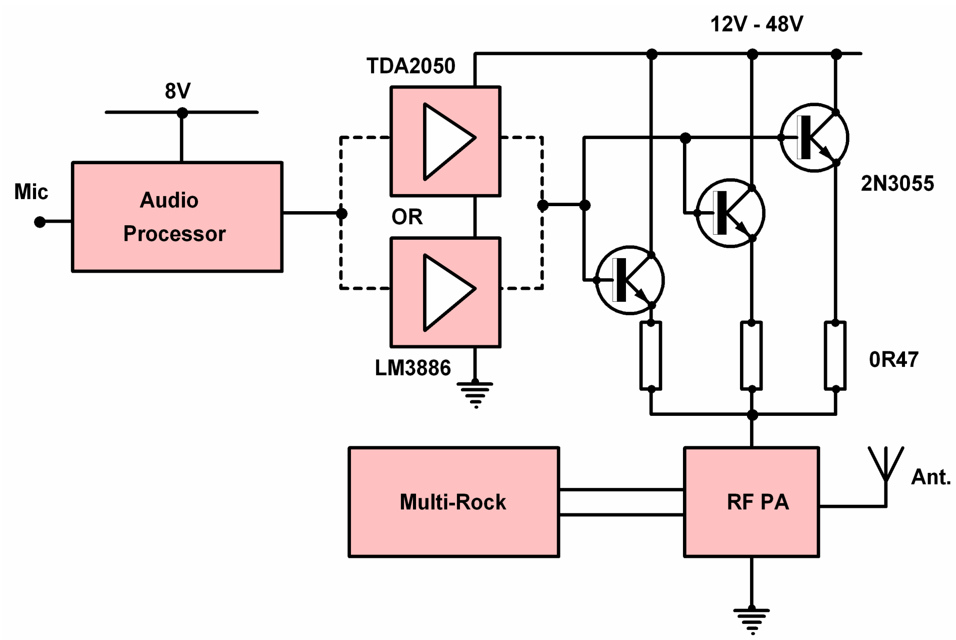

Figure 5.2 FAT5 basic interconnection diagram

Figure 5.2 above shows how the modulator sections connect together to form a complete series modulator for the FAT5 transmitter. It should be apparent from the diagram that the RF PA is really just the emitter load resistor in a common emitter audio amplifier. Appreciation of this helps when deciding how to connect earth leads from various parts of the circuit. Keep low level earth current routes e.g. mic input away from high current routes e.g. PA.

- Decouple the composite PA feed from the emitter resistors with a 100n to earth (not shown in above diagram).
- An RF choke may be needed here if any RF issues are apparent in the modulator. This is generally not required. Use a few turns of wire through a ferrite core.

5.5 Setting the Modulation level

There are three potentiometers to adjust:

1. RV1 on the speech processor PCB. This is the compressor drive, which alters the amount of compression. (The output from the audio processor PCB is a fixed level)

   The compression level / mic gain can be adjusted to taste depending on the amount of compression required, and type of microphone used. Turning the control ever higher won't result in a louder signal, just more background noise and a pumping effect on the audio. See section 2.9 for advice on setting up the correct compression level.

2. RV1 on the modulator PCB (either type). This is used for setting modulation depth.

3. RV2 on the modulator PCB (either type). This sets the standing voltage applied to the RF-PA and the correct setting is important for low modulation distortion. Sections 3.2 and 4.2 above covered how to adjust this voltage level when testing the modulator PCB. When setting up a complete system with a correctly tuned PA as a load and the FAT5 audio processor connected, RV2 should be re-adjusted.

   Whilst transmitting (into a dummy load!) apply an audio tone to the mic input of the speech processor PCB, then turn up RV1 on the modulator PCB, observing the modulated PA voltage on an oscilloscope. Increase the level until clipping occurs, and adjust RV2 for symmetrical clipping. After setting RV2, back off RV1 (modulator) to reduce the modulation to about 70%.

5.5.1 Final modulation adjustment

Please note: The output level from the speech processor when driven by a single tone (or a whistle) will be lower than the output level with normal speech. This is a characteristic of this type of compressor. Adjusting the mod level to 100% on a tone then switching to normal voice will result in overmodulation.

To correctly set the modulation depth use a modulation meter, preferably in conjunction with an oscilloscope set to display the modulation envelope (i.e. timebase speed adjusted to audio not RF). Avoid using just a fly-lead connection – phasing effects from multiple RF pickup points can result in misleading readings.

Using normal speech, adjust the mod level control (RV1 on modulator PCB) to give 90-95% peak indication of the mod meter. Check that peaks are not overmodulating by observing the 'scope trace and checking no bright spots are visible at the centre of the trace.

Avoid pushing the last percentile out of the modulator. The modulation will be much cleaner when set at the 95% level as transients from the processor will always be present. (All but the most complex hardware-based speech compressors have overshoot as a high level needs to be present before a level control signal can be gererated to turn it down again). There is no audible difference between a 95% and 100% modulated signal - the level change is too small. Even 90% to 100% is barely perceptible.

Again (and no apologies for repeating this), once the modulation depth has been set on speech, you will only be able to modulate to about 70% with a whistle. This is normal.

6. POWER SUPPLIES

Using the correct power supplies with adequate current available is key to making the project a success. Don't skimp and use something that is barely up to the job. The author recommends commercially available switch-mode power supplies. These are available for very reasonable prices and unless you're very unlucky you will find them RF quiet and totally reliable. They also have the advantage of built-in overload protection/current limiting. There is no reason why less efficient linear supplies cannot be used as long as they are regulated, or the on/off load performance of non-regulated supplies is well understood.

There are on-board regulators on the Audio Processor and Multi-Rock PCBs allowing a single 12V PSU or the shack 13.8V PSU to be used.

The RF_PA requires a stabilised 500mA 8V supply to power the FET drivers. This can be derived from say a 7808 regulator bolted to a small heatsink, or the chassis.

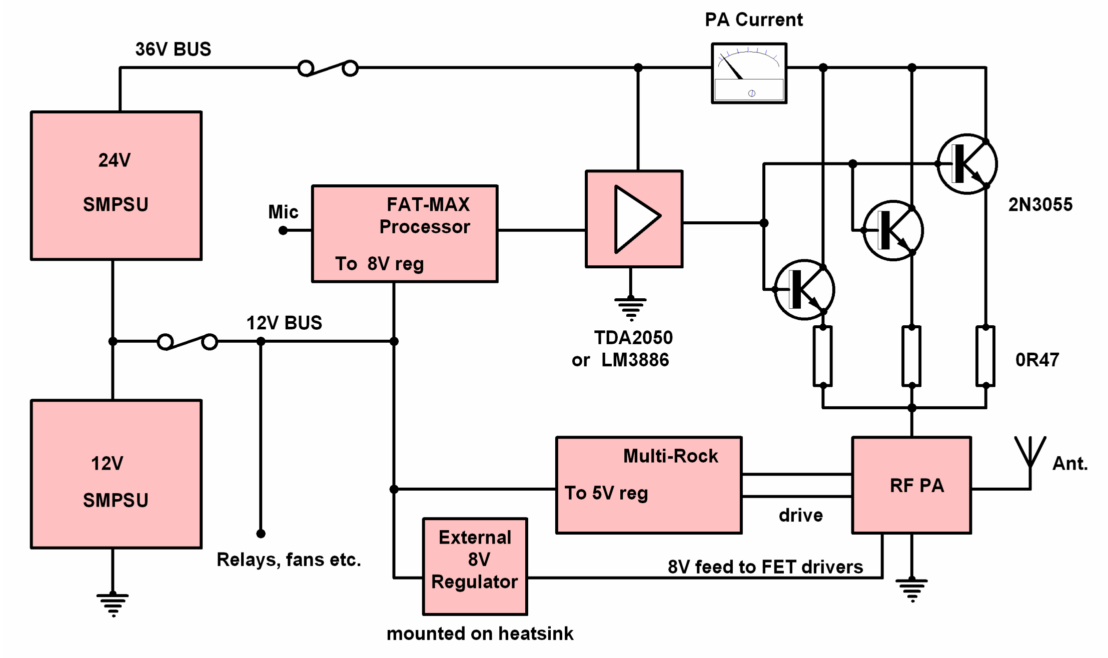

Figure 6.1 Example of PSU implementation using SMPSUs

Firgure 6.1 above shows a way of deriving all voltages from a pair of SMPSUs. They should be rated at 8A+ to comfortably handle peak currents during modulation. The 12V bus feeds the on-board regulators of the FAT-MAX processor and the Multi-Rock signal source. It was originally planned to use a common 8V regulator for FAT-MAX and the RF-PA drivers. **This is no longer recommended.** Separate regulators should be used to remove any chance of transients from the FAT-MAX DC-DC converter modulating the FET drivers.

At the time of writing eBay offers a 24V 10A SMPSU from China for £30.99 including delivery. (item # 260395725583, seller: daifengsu). UK sources of similar units can be found at higher cost. This particular PSU has been tested by GW8LJJ and is RF quiet.

Also currently available eBay is a 12V 10A SMPSU from Hong Kong for £16.50 including delivery (item # 390183049489, seller: chevrolettong). This unit has not been tested although the spec. looks fine.

These SMPSUs tend to mention 'For radio and Television' in the text, and should mention EMI filtering in the specification.

There is no reason in principle why a 24V PSU cannot be piggy-backed onto the shack 12V (13.8V) PSU as long as the 24V PSU output terminals are floating (normally the case, but check first!).

With the voltages shown in the illustration the transmitter will easily provide much more than 50W carrier. To reduce power turn the volts down a little (most PSUs can be adjusted by +/-10%) or add a couple more turns to the PA coil to reduce the on-tune PA current and bring the power back to around 50W. There is a good safety margin at the 50W level – don't erode it for the sake of a fraction of an S-point!

Please contact the author if you have any concerns or questions regarding power supplies.

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Please report any errors/omissions/typos