SPRING REVERBERATION UNIT

by Nicholas Vinen

Here's a blast from the past! Welcome to our *Spring Reverberation Unit* project for musicians. It uses an affordable, readily available spring 'tank' and a flexible power supply, so you can easily build it into your favourite amp – even if it's portable.

Despite the availability of digital reverb and effects units, many musicians, especially guitarists, still like the 'old school sound' of spring reverberation.

Put simply, a reverberation effects unit takes the dull sound of an instrument (including the human voice) being played in a 'dead' space and adds lots of little echoes.

These simulate what it sounds like to perform in an acoustically complex space such as an auditorium, which has lots of different hard surfaces for sound waves to reflect off, making for a much more 'live' sound.

Even if you're playing in a decent hall, adding extra reverb can make the hall sound bigger and grander. It's also a great way to help a beginner musician sound more professional.

To simulate all these acoustic reflections, rather than using digital processing, a spring reverb uses a spring 'tank' comprising two or more actual springs.

Sound waves are generated at one end of the springs using a voice coil, much like a tiny speaker, and just as sound waves travel through air, they will also happily travel down the metal springs. They are picked up at the other end by what is essentially a microphone.

Only, because of the (for lack of a better word) springiness of the springs, and the way they are suspended at either end, the audio signal doesn't just travel down the springs, it bounces around, generating echoes and since no physical process is 100% efficient, these decay, sound just like waves do when they bounce off walls, floors, ceilings, chairs and other objects.

It's a personal preference, but many prefer this effect to a digitally generated one.

The end result is something you really have to hear to appreciate, but it's surprising just how good a job the spring tank does of mimicking sounds bouncing around a hall.

Of course, the exact sound depends upon the exact tank used – some have two springs, some have three, some are longer or shorter and so on – but regardless of how natural it is, chances are you will find some configuration where it will add an extra dimension to your performance.

And being electronic, you can vary the reverb effect's intensity (or 'depth') and turn it on or off as necessary. But unlike a digital effects unit, you can't easily change other parameters such as the echo delay or frequency response.

Sourcing the spring tank

Fortunately, there are multiple suppliers of spring reverb tanks. You guessed it; most of them seem to be in China.

The one we're using is from a musical instrument component supplier called Gracebuy based in Guangdong, and at the time of writing this, you could purchase the tank for US$20 including free postage via the following 'shortlink': **http://bit.ly/2GvH9HG**

The same supplier sells this same unit on **ebay.co.uk**. Look for item 301900482233, or search for: 'Spring Reverb Tank Electric Guitar Amplifier 2 Spring Medium Decay TPSB2EB2C1B'.

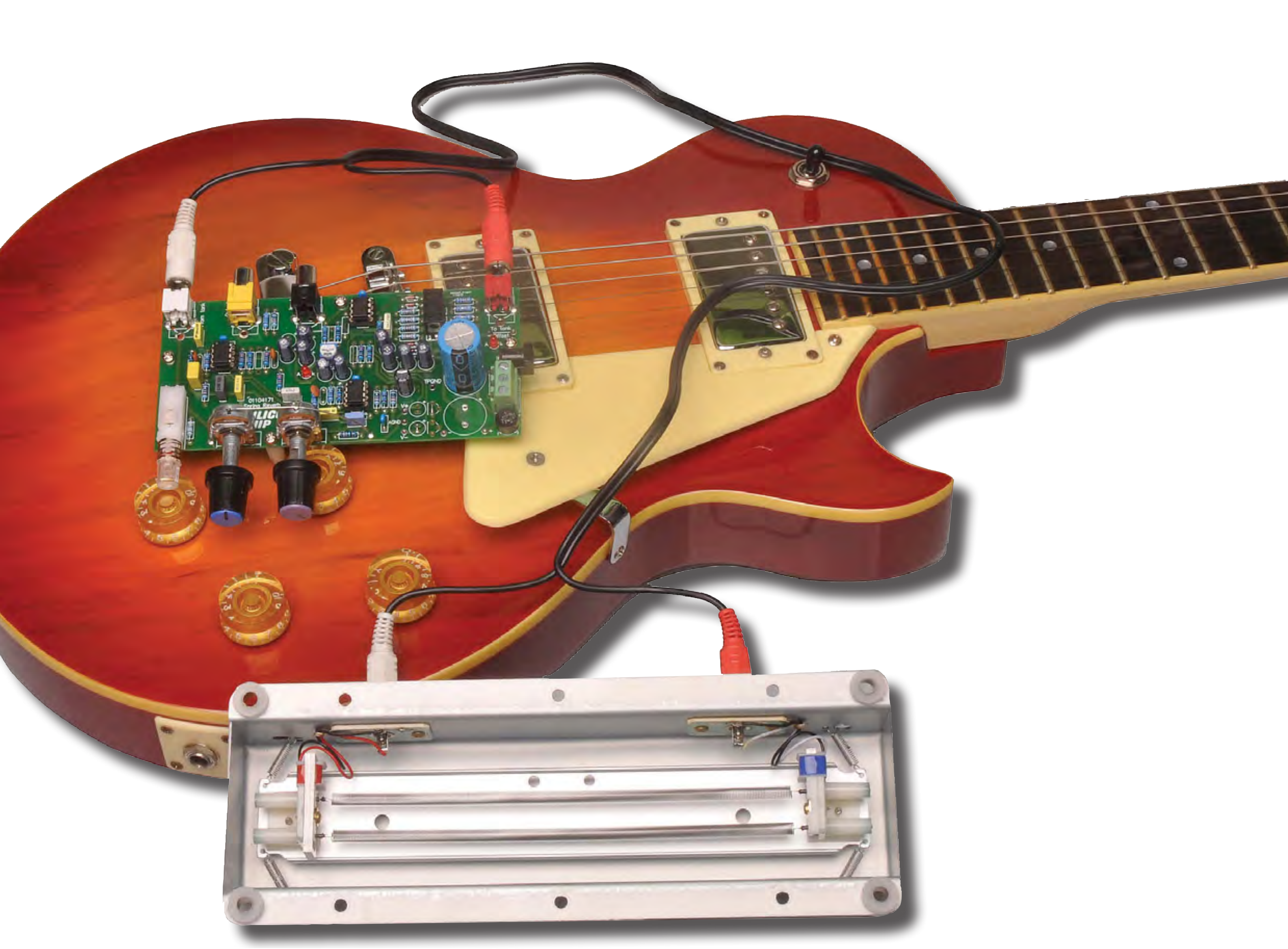
If you search eBay, you can also find other units, including some with three springs and/or longer springs. We haven't tried any of these but we would expect them to work with our circuit with little or no modification. So if you're feeling adventurous, here is a couple of eBay numbers for alternative spring tanks: 391937355232, 162782505026.

You can get an idea of the properties of the tank we're using by looking at the scope screen grabs in Figs.4-7. Three spring units will have triplets of echoes, rather than pairs, and longer units will have a larger gap between the stimulus and echo. Other tanks may also have a shorter or longer persistence time than the one we've used, depending on the properties of the springs themselves.

Note that most of the alternative tanks are larger than the one we've used (which is fairly compact; see the specifications panel) so make sure you have room for it in your amplifier's chassis (or wherever you plan to fit it) before ordering one.

Features and specifications

Reverb tank type: two spring
Anti-microphonic features: spring suspension, plastic mounting bushings
Spring tank dimensions: 235 × 87 × 34mm
Reverb delay times: 23ms, 29ms (see Figs.4 and 5)
Reverb decay time: around two seconds (see Fig.6)
Input sensitivity: ~25mV RMS
Frequency response (undelayed signal): 20Hz-19kHz (–3dB) (see Fig.2)
Frequency response (reverb signal): 200Hz-3.4kHz (–3dB) (see Fig.2)
Signal-to-noise ratio (undelayed signal): 62dB
Signal-to-noise ratio (typical reverb setting): 52dB
THD+N (undelayed signal): typically around 0.05% (100mV signal)
Controls: level, reverb depth, reverb on/off
Power supply: 9-15VAC, 18-30VAC centre tapped or 12-15V DC
Quiescent current: typically 30-40mA



You might think of it as 'olde world' but there's a surprising number of musos who say that a spring reverb always sounds better than a digital unit!

Improvements to the design

Compared to many other designs, an important aspect of this design is its ability to run off a DC supply. This was added so that buskers can add a spring reverb function to portable amplifiers, which may be powered by a 12V lead-acid battery or similar. In fact, the PCB is quite flexible and can be powered from 9-15VAC, 18-30VAC (centre-tapped) or 12-15V DC.

It's also possible to modify it to run off 15-30V DC, in which case you may need to increase the voltage ratings of the 1000µF and 220µF capacitors.

One small extra feature we've added, besides the new power supply options and related changes, is an indicator LED to show whether the reverb effect is active. It's built into the reverb on/off pushbutton switch, S1.

Basic concept

A block diagram of the *Spring Reverb Unit* is shown in Fig.1. The level of the incoming signal (from a guitar, keyboard, microphone, preamp...) is adjusted using potentiometer VR1 and is then fed both to a preamplifier for the spring tank and to a mixer, which we'll get to later. The preamplifier boosts high frequencies since the transducer which drives the springs is highly inductive and so needs more signal at higher frequencies to produce sufficient motion in the springs.

Between the preamp and the tank is the buffer stage, which has little gain but serves mainly to provide sufficient current to drive the transducer, which it does in bridge mode, for reasons explained below.

The output of the spring tank, which is delayed compared to the input and contains all the added reverberations, is fed to switch S1 which can shunt the signal to ground if reverb is not currently required. Assuming the signal is not shunted, it is fed to a recovery amplifier that boosts its level back up to a similar level to the input signal and then on to VR2, which is used to attenuate the reverberations in order to control the intensity or 'depth' of the effect.

The attenuated reverberations are then fed to the mixer, where they are mixed with the clean input signal to produce the final audio output, which can then be fed to an amplifier or mixer.

Circuit description

The complete circuit for the *Spring Reverb* module is shown in Fig.3. Note that two different ground symbols are used in the circuit. For the moment, you can consider them equivalent; we will explain the significance later, when we go over the power supply details.

The signal from the guitar/preamp/source is applied via RCA connector CON1 and then passes through a pair of electrolytic capacitors connected back-to-back (ie, in inverse series), which effectively form a bipolar electrolytic capacitor, to prevent any DC component of the signal from reaching the rest of the circuitry.

The signal then goes through a low-pass/RF filter comprising a 100Ω resistor, 4.7nF MKT capacitor and a ferrite bead. The –3dB point of the low-pass filter is around 340kHz, while the ferrite bead helps attenuate much higher

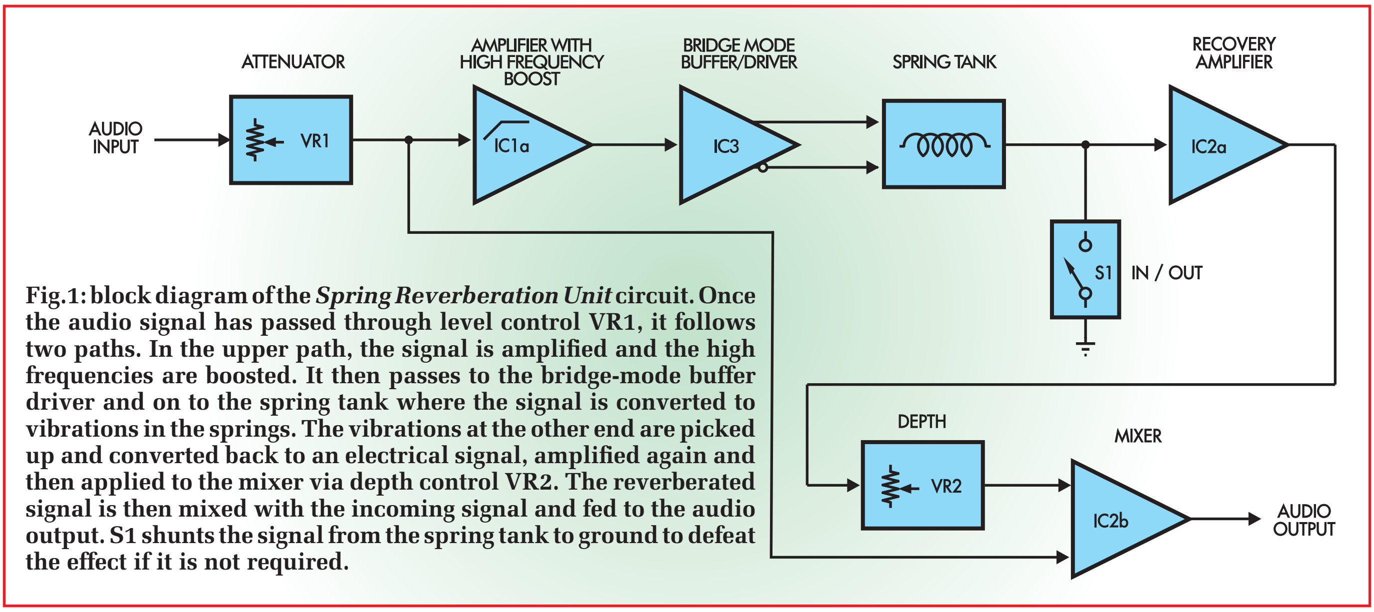


Fig.1: block diagram of the *Spring Reverberation Unit* circuit. Once the audio signal has passed through level control VR1, it follows two paths. In the upper path, the signal is amplified and the high frequencies are boosted. It then passes to the bridge-mode buffer driver and on to the spring tank where the signal is converted to vibrations in the springs. The vibrations at the other end are picked up and converted back to an electrical signal, amplified again and then applied to the mixer via depth control VR2. The reverberated signal is then mixed with the incoming signal and fed to the audio output. S1 shunts the signal from the spring tank to ground to defeat the effect if it is not required.

frequency signals (eg, AM and CB radio) which may be picked up by the signal lead. Both filters help prevent radio signal break-through. The audio signal then passes to 50kΩ logarithmic taper potentiometer VR1, which forms an input level control.

The level-adjusted signal from the wiper of VR1 goes to two different parts of the circuit, as shown in the block diagram (Fig.1); to the mixer, via a 47nF AC-coupling capacitor and to the tank drive circuit, via a 100nF AC-coupling capacitor. We'll look at the latter path first before coming back to the mixer later. The 100kΩ DC-bias resistor at input pin 3 of IC1a forms a high-pass filter in combination with the 100nF coupling capacitor, which has a –3dB point of 16Hz.

Note that in the original design, this part of the circuit used a 10nF capacitor which gave a –3dB point of 160Hz. The reason for having such a high roll-off was two-fold: first, the tank used previously had a very low input DC resistance and presenting it with a high-amplitude, low-frequency signal risked overloading the driving circuitry. And second, this helped attenuate 50/100Hz mains hum and buzz that may be from the guitar, cabling and so on.

Additionally, while it is possible to get good low-frequency performance, it's generally undesirable because it tends to muddy the sound.

We've shifted this –3dB point down because the transducer in the tank we're using this time has a much higher DC resistance and we've beefed up the driving circuitry, so overload is less of a problem, and this makes the reverb sound less 'tinny'.

However, you still have the option of reducing this capacitor value, possibly back to the original 10nF, if you find the unit has excessive hum pick-up. It really depends on your particular situation whether this is likely. Note that this solution to hum is a case of 'throwing the baby out with the bathwater'; at the same time as reducing the hum pick-up, you're also filtering out any genuine signals at similar frequencies.

Getting back to the signal path, IC1a operates as a non-inverting amplifier with a maximum gain of 101, as set by the ratio of the 100kΩ and 1kΩ resistors.

The 10nF capacitor in series with the 1kΩ resistor causes the resistance of the lower leg of the voltage divider to increase at lower frequencies, thus reducing the gain at lower frequencies. For example, a 10nF capacitor has an impedance of 16kΩ at 1kHz, thus the gain at 1kHz is reduced to 100kΩ ÷ (1kΩ + 16kΩ) + 1 = 6.9. The slope of the resulting filter is 6dB/octave and the –3dB point is 16kHz, which not coincidentally, happens to be the frequency at which a 10nF capacitor has an impedance of 1kΩ. In other words, the gain is reduced to about half its maximum (ie, 51) at 16kHz. You can see the effect of this filter stage in the frequency response diagram of Fig.2.

This 10nF capacitor also prevents the input offset voltage of IC1 from being amplified and creating a large DC offset at the output, while the 100pF capacitor across the 100kΩ resistor reduces the gain of this op amp stage at very high frequencies, preventing instability and also reducing the effect of RF/hum pick-up in the PCB tracks. The –3dB high-frequency roll-off point due to this capacitor is 16kHz.

Tank drive circuitry

Because the spring tank we're using has a fairly high input impedance of 600Ω at 1kHz, and because the springs themselves are quite lossy, the signal fed to the tank needs to have as large an amplitude as we can provide, given the supply rails available.

Note that the supplier lists the tank input DC resistance as 28Ω and its inductance as 23mH, but the actual measured figures are 75Ω and 83mH, giving an input impedance of just under 600Ω at 1kHz.

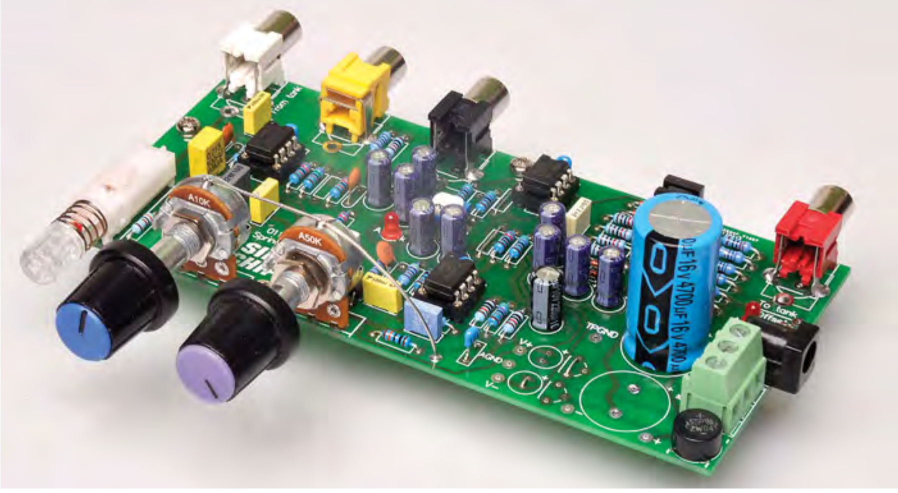
With ±15V supply rails, the LM833 and TL072 low-noise op amps we're using have a maximum output swing of around ±13.5V or 9.5V RMS. But since we've also designed this unit to be able to run off a 12V lead-acid battery (or equivalent) for busking purposes, and with a supply of only 12V, the output swing is much more limited at 9V peak-to-peak or just 3.2V RMS.

To improve this situation, we've redesigned the circuitry to drive the tank in bridge mode. This is possible since the driving transducer's negative input is not connected to its earthed chassis. That doubles the possible signal when running from a 12V DC supply, to nearly 6.5V RMS.

It works as follows. The output signal from gain/filter stage IC1a passes to both halves of dual op amp IC3. In the case of IC3a, it is fed directly to the non-inverting input at pin 3, while for IC3b, it goes to the inverting input at pin 6 via a 4.7kΩ resistor.

IC3a operates as a unity-gain power buffer. The output signal from pin 1 of IC3a goes to the tip connector of CON2 and hence the transducer in the spring tank via a 220Ω series resistor, but pin 1 also drives the bases of complementary emitter-follower pair Q1 and Q2 via two 22µF capacitors.

A DC bias voltage of around 0.7V is maintained across these capacitors due to the current flowing from the regulated V+ rail (typically +15V),



Here's the completed *Spring Reverberation Unit* (in this case to suit a DC power supply (see Fig.8[a]). Note the tinned copper wire link over the potentiometer bodies – it not only helps minimise hum but also keeps the pots themselves rigid.

through a 2.2kΩ resistor, small signal diodes D1 and D2, another 2.2kΩ resistor and to the V– rail (typically –15V). You can calculate the current through this chain at around (30V – 0.7V × 2) ÷ (2.2kΩ × 2) = 6.5mA, and this current sets the forward voltage across D1 and D2 and thus the average voltage across those two capacitors.

The voltage across these capacitors defines the quiescent base-emitter voltage of both Q1 and Q2, and thus their quiescent current, which is around 10mA. This is necessary to prevent significant crossover distortion when drive is being handed over between Q1 and Q2, as the output signal passes through 0V.

The two 10Ω emitter resistors help to stabilise this quiescent current by way of local negative feedback, since as the current through Q1 or Q2 increases, so does the voltage across these resistors, which reduces the effective base-emitter voltage.

The signal fed to the tank is also fed back to inverting input pin 2 of IC3a, setting the gain of this stage at unity. This closes the op amp feedback loop around Q1, Q2 and associated components.

The outer 'ring' terminal of CON2, which connects to the opposite end of the tank drive transducer, is driven by an almost identical circuit based on IC3b and transistors Q3 and Q4. However, so that the transducer is driven in bridge mode, the gain of this stage is –1, ie, it is an inverting unity-gain amplifier.

This is achieved by connecting its pin 5 non-inverting input to signal ground via a 2.2kΩ resistor and then using a 4.7kΩ feedback resistor and a 4.7kΩ resistor between the inverting input (pin 6) and the output of the previous stage, pin 1 of IC1a. The 2.2nF feedback capacitor rolls off the gain of this stage at high frequencies, giving a –3dB point of 16kHz and ensuring stability. The tank doesn't do much to preserve frequencies above 5kHz anyway.

By the way, we're using a TL072 op amp for IC3 instead of an LM833, as used for IC1 and IC2, because its lower bandwidth (and other aspects of the internals of this IC) makes it better suited for driving a complementary emitter-follower buffer. If you use an LM833 instead, the circuit will work but there is likely to be a spurious low-level ~1MHz signal injected which might upset the power amplifier.

This signal is due to the op amp having trouble coping with the extra phase shift introduced due to the transistors in its feedback path and it's hard to tame without adding some gain to the buffer stage, which we don't really need. Using a TL072 instead solves the problem and since all the gain is handled by the other two LM833 op amps (which have a lower noise figure), it doesn't degrade the performance at all.

Output offset adjustment

Since the transducer in the tank has a relatively low DC resistance, we'd like to avoid a high DC offset voltage across CON2 as this will waste power and heat up both the transducer and Q1-Q4 unnecessarily. It's not absolutely critical, but we've included DC offset adjustment circuitry because it's relatively simple and cheap.

But because this unit can run off an unregulated DC supply, we've designed it so that it doesn't rely on the regulated supply rails to provide a consistent offset adjustment.

Red LED1 and LED2 are connected across the supply rails with 4.7kΩ current-limiting resistors. The junction of LED1's cathode and LED2's anode is connected to signal ground. As a result, LED1's anode is consistently around 1.8V above signal ground while LED2's cathode is consistently about 1.8V below signal ground.

VR3 is connected between these two points and so the voltage at its wiper can be adjusted between these two voltages. Two back-to-back 22µF capacitors stabilise this voltage so that it does not jump around when power is first applied and the supply rails are rising. A 470kΩ resistor between VR3's wiper and pin 2 of IC1a allows VR3 to slightly increase or decrease the voltage at that pin, to cancel out any offset voltages in op amps IC1a, IC3a and IC3b.

Note that because IC3a has a gain of +1 and IC3b has a gain of –1, when you turn VR3 clockwise, the output voltage of IC3a will rise slightly while the output voltage of IC3b will drop slightly. Thus, there will be a position of VR3 such that the output voltages of these two op amps are identical when there is no input signal. This is the condition we're aiming for as it minimises DC current flow through the transducer connected to CON2.

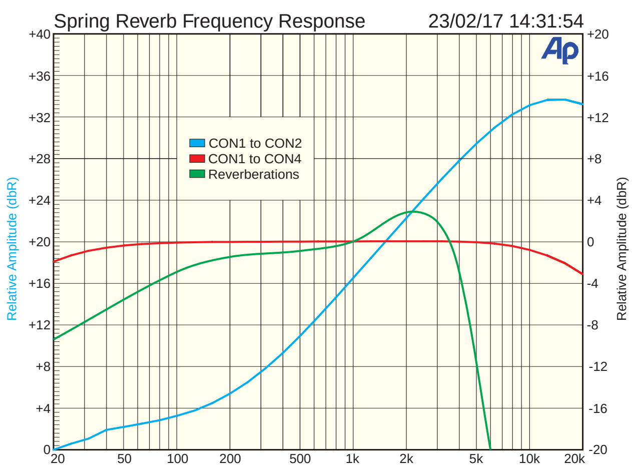
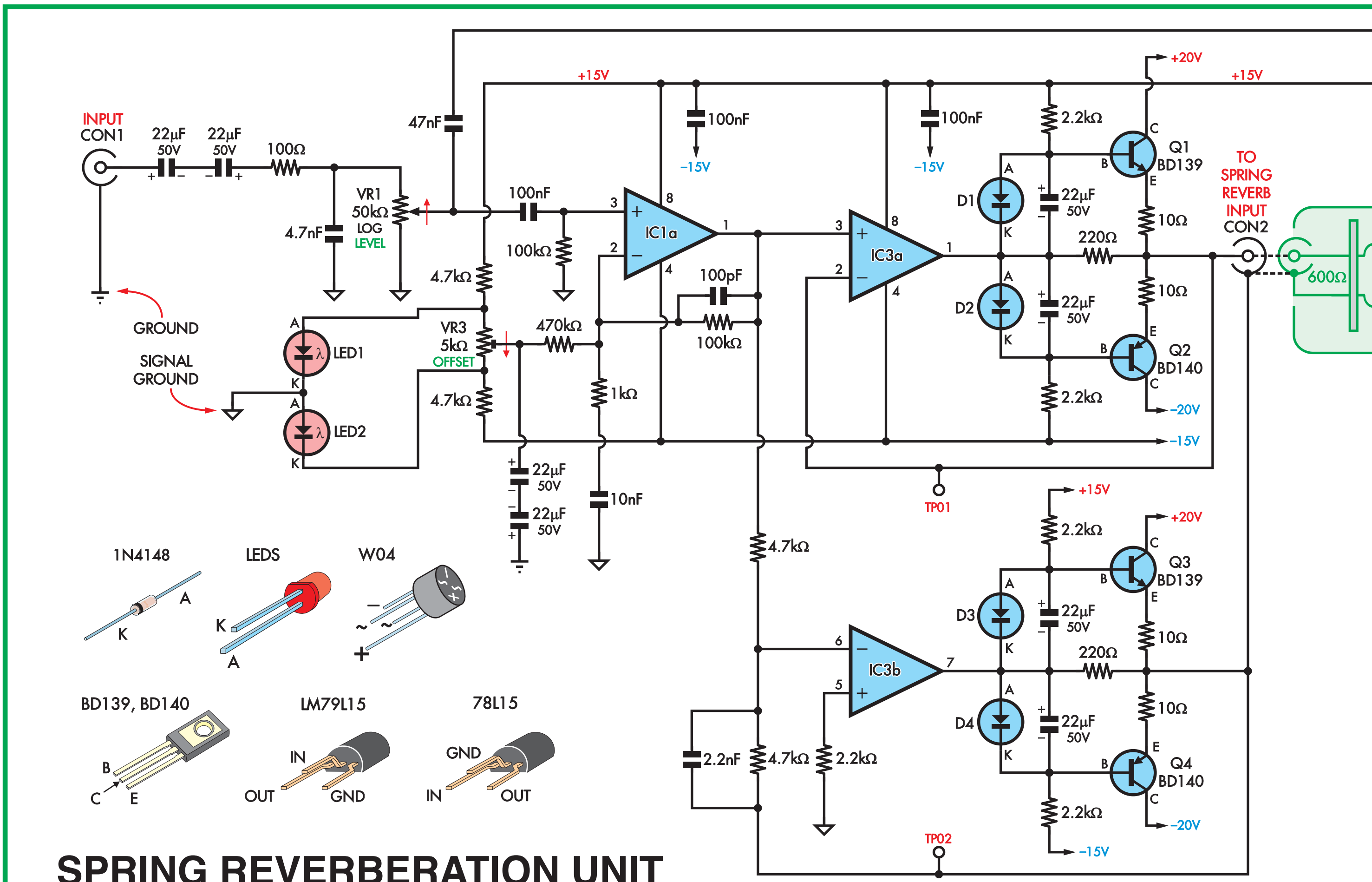


Fig.2: three frequency response plots for the *Spring Reverberation Unit*. The frequency response from input connector CON1 to spring tank driver connector CON2 is shown in blue and uses the left-hand Y-axis. The unit's overall frequency response, ignoring reverberations, is shown in red. The approximate frequency response for the reverberations is shown in green. This is difficult to measure since pulse testing must be used, otherwise standing waves cause constructive / destructive interference. Our curve is based on pulse testing at discrete frequencies and can be considered an approximation of the actual response.



SPRING REVERBERATION UNIT

Fig.3: complete circuit for the *Spring Reverberation Unit*, including the spring tank connected between CON2 and CON3 (shown in green). Only the output socket of the spring tank is connected to its case – this is to avoid earth (hum) loops. Note also that two different ground symbols are used; depending on the power supply arrangement, they may be connected together, or the signal ground may sit at half supply when powered from DC. Two different power supply arrangements are shown in the boxes at right and the PCB can be configured for one or the other. With an AC input, the circuit is powered from regulated, split rails of nominally ±15V while with a DC supply, the circuit runs off the possibly unregulated input supply.

Table 1 – expected voltages relative to TPGND

Supply	'+'	'–'	V+	V-	AGND
15VAC	+20V	-20V	+15V	-15V	0V
12VAC	+17V	-17V	+12V	-12V	0V
9VAC	+12V	-12V	+9V	-9V	0V
12V DC	+12V	0V	+12V	0V	+6V *(half V+)*

Signal recovery

The signal passes through the springs in the tank as longitudinal vibrations; and these are picked up at the opposite end by another transducer which is connected to the board via CON3. The signal from this second transducer is roughly –60dB down compared to the signal going in, so it is fed to another high-gain stage based around op amp IC2a, through another coupling/high-pass filter comprising a 100nF capacitor and 100kΩ resistor, with a –3dB point of around 16Hz.

Switch pole S1d is shown in the on position; in the off position, it shorts the signal from the tank to ground, so there is effectively no reverb.

IC2a is configured as a non-inverting amplifier with a maximum gain of 83 (820kΩ ÷ 10kΩ + 1). However, like IC1a, its gain is reduced at lower frequencies due to the 15nF capacitor in the lower leg of the divider, with a –3dB point of around 1kHz. As before, a capacitor across the feedback resistor ensures stability and reduces gain at very high frequencies; in this case, it is 10pF.

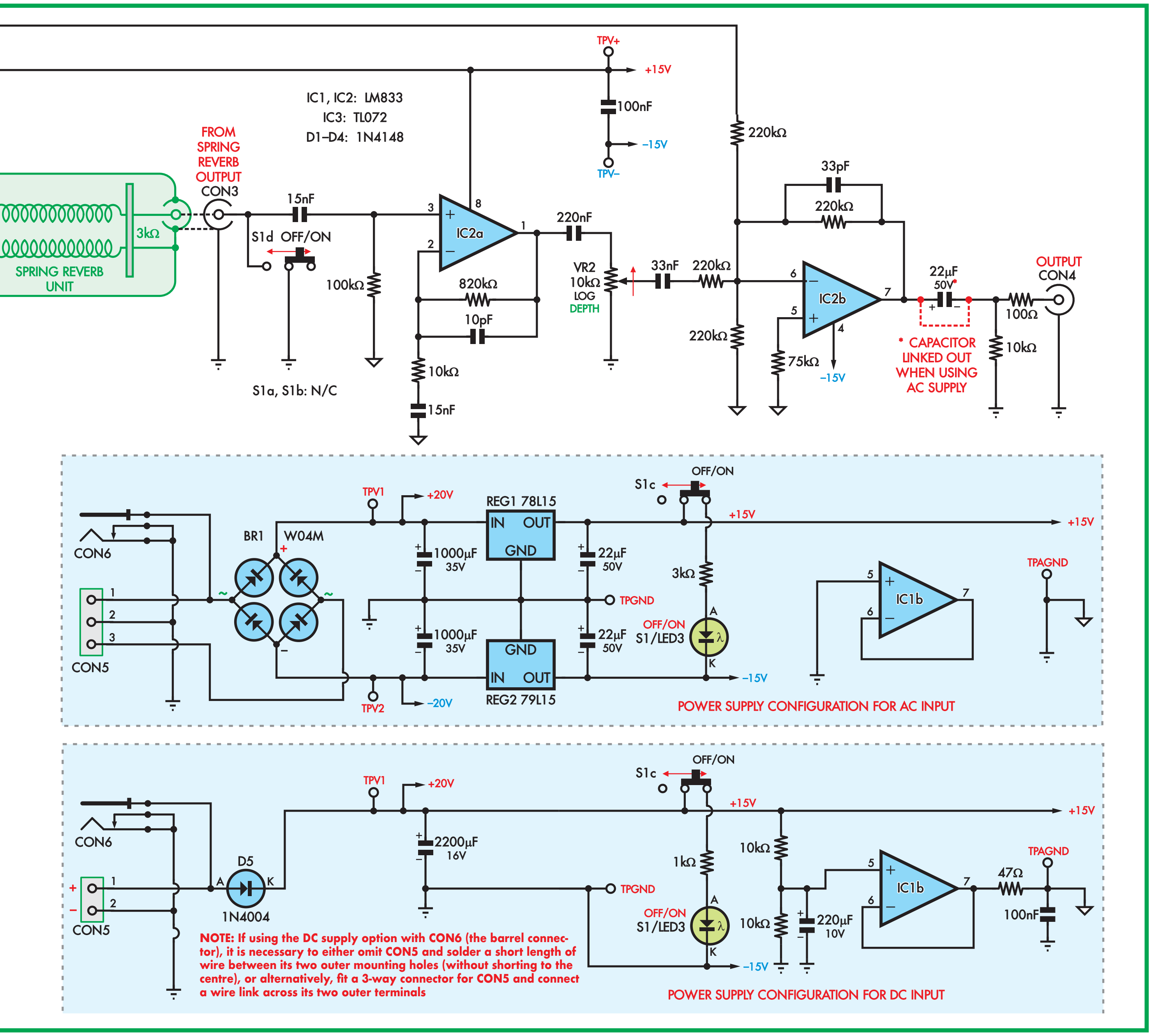
The recovered signal from the tank is then AC-coupled to 10kΩ log potentiometer VR2 via a 220nF capacitor. VR2 controls the level of the reverb signal which is fed to the mixer, and thus the 'depth' of the reverb effect. The resulting signal at its wiper is then coupled to inverting pin 6 of mixer op amp IC2b via a 33nF AC-coupling capacitor and 220kΩ series resistor.

The reason for using two coupling capacitors with VR2 is to prevent any DC current flow through it, which could cause crackling during rotation as the pot ages (note that we have done the same with VR1).

The mixer

You may remember that the signal from VR1 was fed both to the tank and to the mixer; after being coupled across the 47nF capacitor, it passes through a second 220kΩ series resistor to also reach pin 6 of IC2b. So this is the point at which the original and reverberated signals meet and you can see how VR2 is used to vary the effect depth, as the louder the reverb signal is compared to the input signal, the more reverberation will be evident.

A third 220kΩ resistor provides feedback from IC2b's output pin 7 back to its inverting input, while the non-inverting input (pin 5) is connected to signal ground via a 75kΩ resistor. This value was chosen to be close to the value of three 220kΩ resistors in parallel, so the source impedance of both inputs is similar. IC2b operates as a 'virtual earth' mixer, with both



its input pins 5 and 6 held at signal ground potential.

Remember that the action of an op amp is to drive its output positive if the positive input is higher than the negative input and negative if the situation is reversed. So the feedback from its output to its inverting input operates to keep both inputs at the same potential. Since the non-inverting input is connected to ground, the inverting input will be held at that same potential and the signals represented by the currents flowing through the three 220kΩ resistors are mixed and appear as an inverted voltage at the output.

The output of IC2b is fed to output RCA connector CON4 via a 22µF AC-coupling capacitor and 100Ω short circuit protection/stabilisation resistor. The capacitor removes the DC bias from the output when a DC power supply is used. If an AC supply is used, the output of IC2b will already swing around 0V so no DC-blocking capacitor is needed and it is linked out.

Note that the PCB has provision for two back-to-back electrolytics here (for use with an AC supply). However, IC2b's output offset should be low enough that most equipment that would follow the reverb unit (eg, an amplifier) should not be upset by it, hence we are not recommending that you fit them.

Power supply

Two different configurations for the power supply are shown in Fig.3 and you can choose one or the other depending on which components you fit. The one at top suits a transformer of 9-15VAC (or 18-30VAC centre tapped). AC plugpacks can be used. The power supply configuration at bottom is intended for use with 12V batteries or DC plugpacks and will run off 12-15V DC. However, it could easily be adapted to handle higher DC voltages of up to 30V if necessary.

Looking at the AC configuration at top, the transformer is normally wired to CON5. If it isn't centre tapped, the connection is between pin 2 and either pin 1 or pin 3. For tapped transformers, the output is full-wave rectified by bridge rectifier BR1, while for single windings, the output is half-wave rectified. The output from BR1 is then fed to two 1000µF filter capacitors and on to linear regulators REG1 and REG2, to produce the ±15V rails.

If your AC supply is much lower than 15V (or 30V centre tapped), you will need to substitute 78L12/79L12 regulators for REG1 and REG2 to prevent ripple from feeding through to the output. Similarly, for AC supplies below 12V (or 24V centre tapped), use 78L09/79L09 regulators.

Fig.4: the yellow trace shows the signal fed to the spring tank input, while the green trace at bottom shows the signal at the spring tank output. 23.6ms after a pulse is applied to the input, it appears at the output and then a second echo appears around 29ms after the initial pulse. You can see the next set of echoes due to the signal travelling up and down the springs again some 45ms later and note that each set of echoes has opposite polarity compared to the last.

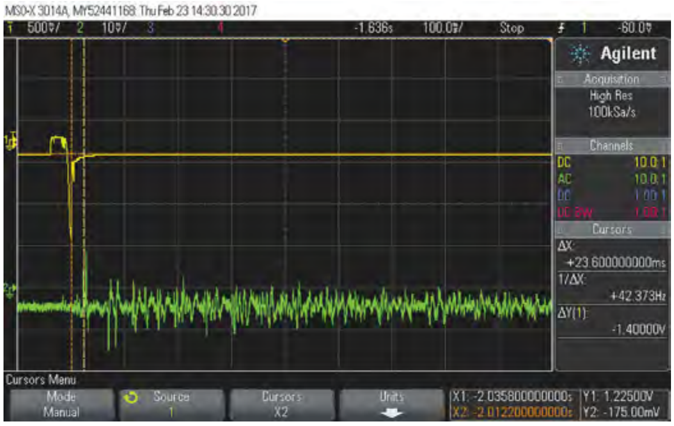


Fig.5: the same signal as shown in Fig.4 but this time at a slower timebase, so you can see how the reverberating echoes continue on for some time after the initial pulse, slowly decaying in amplitude.

Assuming the reverb effect is on, switch pole S1c will be in the position shown and so the LED within S1 will be lit, with around 9.3mA [(30V – 2V) ÷ 3kΩ] passing through it.

Op amp stage IC1b is not used with an AC supply and so its non-inverting input is connected to ground and its output to its inverting input, preventing it from oscillating or otherwise misbehaving. With an AC supply, the signal ground is connected directly to the main (power) ground via a link.

DC supply

For a DC supply, such as a 12V battery, the configuration at bottom is used.

If using the DC supply option with CON6 (barrel connector), it is necessary to either omit CON5 and solder a short length of wire between its two outer mounting holes (without shorting to the centre), or alternatively, fit a 3-way connector for CON5 and connect a wire link across its two outer terminals.

Diode D5 replaces the bridge rectifier and provides reverse polarity protection. The main filter capacitor is larger, at 2200µF, to minimise supply ripple.

Parts list – Spring Reverberation Unit

1 double-sided PCB, available from the *EPE PCB Service*, coded 01104171, 142 × 66mm
1 spring reverb tank (see text)
1 stereo RCA lead with separate shield wires
4 RCA sockets, switched horizontal or vertical (CON1-CON4)
1 3-way terminal block, 5.08mm pitch (CON5) OR
1 PCB-mount DC socket, 2.1mm or 2.5mm ID (CON6)
1 50kΩ logarithmic taper single-gang 16mm potentiometer (VR1)
1 10kΩ logarithmic taper single-gang 16mm potentiometer (VR2)
1 5kΩ mini horizontal trimpot (VR3)
2 knobs to suit VR1 and VR2
1 4PDT push-push latching switch with integral LED (S1)
8 PCB pins (optional)
1 100mm length 0.7mm diameter tinned copper wire
3 8-pin DIL sockets (IC1-3) (optional)

Semiconductors
2 LM833 low noise dual op amps (IC1,IC2)
1 TL072 low noise JFET-input dual op amp (IC3)
2 BD135/137/139 1.5A NPN transistors (Q1,Q3)
2 BD136/138/140 1.5A PNP transistors (Q2,Q4)
2 red 3mm LEDs (LED1,LED2)
4 1N4148 signal diodes (D1-D4)

Capacitors
10 22µF 50V electrolytic
1 220nF 63/100V MKT
2 100nF 63/100V MKT
3 100nF multi-layer ceramic
1 47nF 63/100V MKT
1 33nF 63/100V MKT
1 15nF 63/100V MKT
1 10nF 63/100V MKT
1 4.7nF 63/100V MKT
1 2.2nF 63/100V MKT
1 100pF ceramic
1 33pF ceramic
1 10pF ceramic

Resistors (all 0.25W, 1%)

1 820kΩ	1 470kΩ	3 220kΩ	3 100kΩ	1 75kΩ
2 10kΩ	4 4.7kΩ	6 2.2kΩ	1 1kΩ	2 220Ω
2 100Ω	4 10Ω			

Additional parts for 9-15VAC powered version
1 78L09, 78L12 or 78L15 positive 100mA regulator (REG1) (see text)
1 78L09, 79L12 or 79L15 negative 100mA regulator (REG2) (see text)
1 W02/W04 1A bridge rectifier (BR1)
2 1000µF 35V/50V electrolytic capacitors, 16mm maximum diameter, 7.5mm lead spacing
1 22µF 50V electrolytic capacitor
1 3kΩ 0.25W 1% resistor

Additional parts for 12-15V DC powered version
1 1N4004 1A diode (D5)
1 2200µF 16V electrolytic capacitors, 16mm maximum diameter, 7.5mm lead spacing
1 220µF 10V electrolytic capacitor
1 100nF multi-layer ceramic capacitor
2 10kΩ 0.25W 1% resistors
1 1kΩ 0.25W 1% resistor
1 47Ω 0.25W 1% resistor

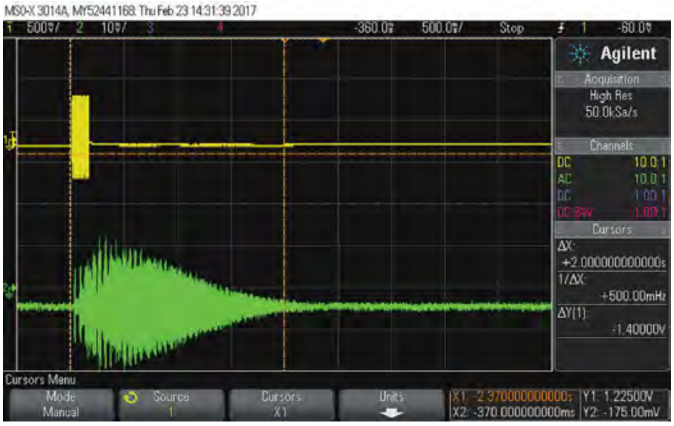


Fig.6: here we have a longer stimulus pulse, again shown in yellow, and the response shown in green on a much longer timebase. The reverberations continue for several seconds after the initial pulse but they have mostly died out after around two seconds (indicated with the vertical cursor).

Fig.7: this shows the output of the *Spring Reverberation Unit* with a short 1kHz burst applied to the input. You can see the original pulse at the left side of the screen and the reverberating pulses, which have been mixed into the same audio signal, repeated twice with decaying amplitude.

For DC supply voltages above 15V, substitute a similarly-sized capacitor with a higher voltage rating such as 2200µF/25V or 1000µF/50V.

The current-limiting resistor for LED3 has been reduced to 1kΩ so that it is still sufficiently bright with the reduced supply voltage, while IC1b is configured to generate a virtual earth at half supply. This is derived from the main supply via a 10kΩ/10kΩ resistive divider with a 220µF capacitor across the bottom leg to eliminate supply ripple from the signal ground.

Op amp IC1b is configured as a buffer, so that the signal ground has a low impedance and drives it via a 47Ω resistor, to ensure op amp stability.

A 100nF capacitor between signal ground and power ground keeps the high-frequency impedance of the signal ground low despite this resistor.

PCB construction

Assembly of the PCB is straightforward. It is available from the *EPE PCB Service*, coded 01104171 and measures 142 × 66mm with tracks on both sides, and plated through-holes. Two overlay diagrams are shown overleaf: Fig.8(a) shows the component layout for a DC supply, while Fig.8(b) shows the layout for an AC supply. Differences between the two will be noted in the following instructions.

Begin by fitting small signal diodes D1-D4, oriented as shown in Fig.8 and then use the lead off-cuts to form the wire links, shown in red. Both versions require five links to be fitted, but some of them are in different places so follow the appropriate overlay diagram.

Next, fit the resistors where shown. While their colour code values are shown in the table overleaf, it's a good idea to check the resistor values with a multimeter before fitting them and remember to slip a ferrite bead over the lead of the 100Ω resistor just above VR1.

The resistors fitted to both versions are almost identical; besides the variation in value of the resistor next to S1, the only other difference is that the three resistors to the right of IC1 are not fitted for the AC supply version.

For the DC supply version, you can now fit D5, oriented as shown.

If you are using IC sockets, solder them in place now, with the notched ends towards the top of the board. Otherwise, solder the three op amp ICs directly to the board with that same orientation. Note that IC3 is a TL072, while the other two ICs are LM833s, so don't get them swapped around.

For the AC supply version, solder BR1 in place with its longer (+) lead towards upper left, as shown in Fig.8.

Now proceed to install the two onboard red LEDs (LED1 and LED2) with the longer anode leads to the left (marked A on the PCB) and all the ceramic and MKT capacitors in the locations shown in the overlay diagram. Polarity is not important for any of these capacitors.

Note that LED1 and LED2 are lit as long as power is applied, so you could mount one of these off-board as a power-on indicator if necessary.

However, we think in most cases, constructors will be building the *Reverb* unit into an amplifier which already has a power-on indicator, so this should be unnecessary and LED1/LED2 can simply be mounted on the PCB as shown.

If you're building the AC-powered version, solder REG1 and REG2 in place now, oriented as shown. Don't get them mixed up. You will probably need to crank out their leads slightly using small pliers, to suit the PCB pads.

Now fit trimpot VR3, followed by illuminated switch S1. Make sure S1 is pushed all the way down onto the PCB before soldering two diagonally opposite pins and then check it's straight before soldering the remaining pins.

You can now install the small (22µF) electrolytic capacitors. These are polarised and the longer (+) lead must go towards the top of the board in each case, as shown using + symbols in Fig.8. If building the DC-powered version, there is also one 220µF capacitor that you can fit at the same time, but make sure it goes in the position indicated.

Next, mount CON5 and/or CON6, depending on how you plan to wire up the power supply. If fitting CON5, make sure its wire entry holes go towards the nearest edge of the board and if using a 2-way connector (for a DC supply), make sure it goes in the top two holes as shown in Fig.8(a).

Next, fit CON1-CON4. In each case, you have a choice of using either a horizontal switched RCA socket (as shown on our prototype) or a vertical RCA socket fitted either to the top or the bottom of the PCB.

Pads are provided for all three possibilities, and which is best depends on how you're planning on running the wiring in your particular amplifier.

As you will see later, we recommend using a stereo RCA-RCA lead to connect the main board to the tank, and the tank will normally be mounted in the bottom of the amplifier chassis while the *Reverb* board will normally be mounted on the front panel. So keep that in mind when deciding which RCA socket configuration to use.

If you want to fit PCB pins for the test points, do so now, however it isn't really necessary since the pads are quite easy to probe with standard DMM leads.

Transistors Q1-Q4 should be fitted next. Don't get the two types mixed up; the BD139s go towards the top of the board, while the two BD140s go below.

All four transistors are fitted with their metal tabs facing towards the bottom of the board as shown; if you're unsure, check the photo.

You can now solder the large electrolytic capacitor(s) in place; the DC supply version has one, located as shown in Fig.8(a), while the AC supply version has two. In all cases, the longer (+) lead goes towards the top of the board, as shown.

The last components to fit to the PCB are potentiometers VR1 and VR2. However, before installing them you must do two things. First, clamp each pot in a vice and file off a small area of passivation on the top of the body, allowing you to solder the ground wire later on.

Second, figure out how long you need the shafts to be to suit your amplifier and cut them to length. Make sure they're still long enough so that you can fit the knobs later!

Now solder the two pots to the board, ensuring that the 10kΩ pot (VR2) goes on the left side and then insert one end of a 100mm length of tinned copper wire in the pad marked 'GND', just to the left of VR2, and solder it in place. Next, bend the wire so it contacts the top of the two pot bodies and then solder it to the free pad to the right, as shown in Fig.8, and trim off the excess.

Now it's just a matter of soldering this ground wire to the areas where you scraped away the passivation from VR1 and VR2. Note that you will need to apply the soldering iron for a few seconds for the metal to get hot enough for solder to adhere.

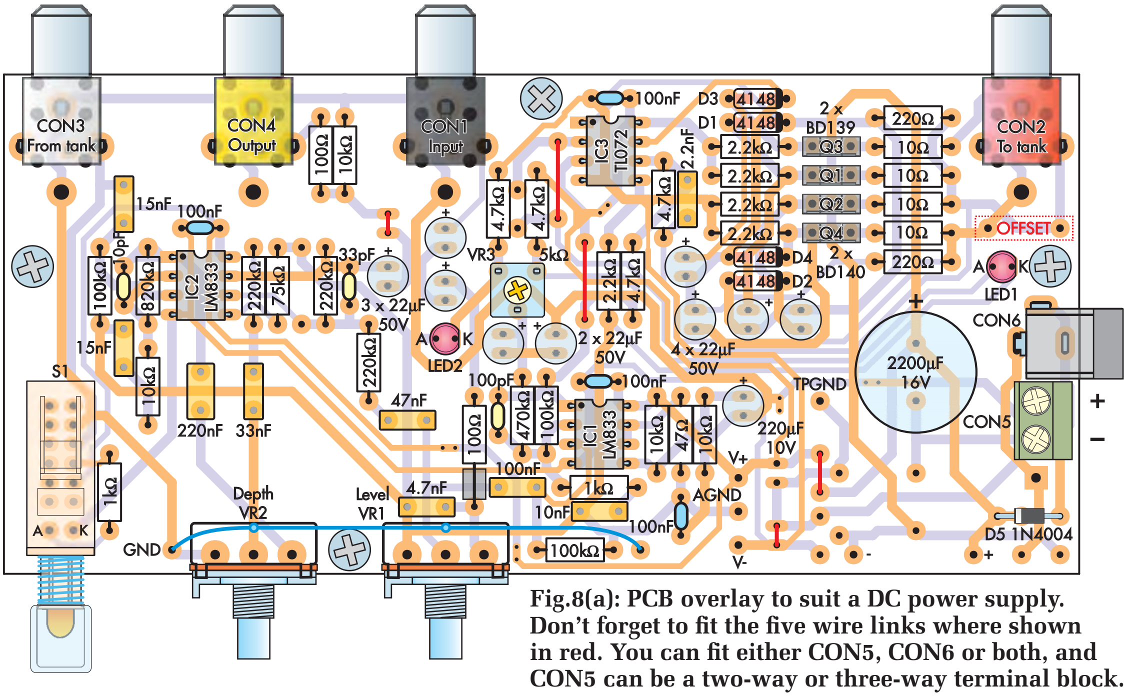


Fig.8(a): PCB overlay to suit a DC power supply. Don't forget to fit the five wire links where shown in red. You can fit either CON5, CON6 or both, and CON5 can be a two-way or three-way terminal block.

Testing and set-up

The first step is to apply power and check the supply voltages. If you've fitted sockets, leave the ICs off the board for the time being. Having said that, if you have configured the board for a DC supply, plug in LM833 op amp IC1 (taking care with its orientation).

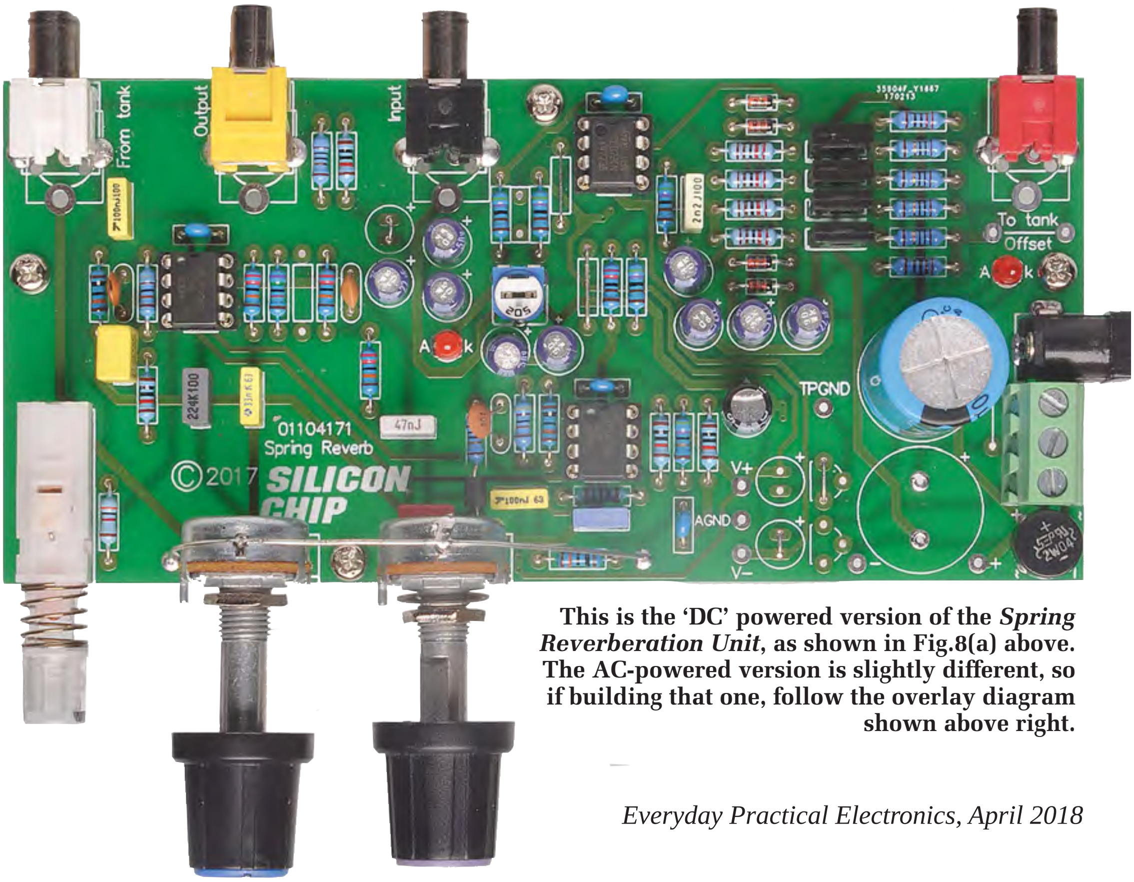
Apply power and check that the voltages at the five specified test points are close to the values given in Table 1 (on the circuit diagram).

Voltage variation on the '+' and '–' test points can be expected to be fairly large, possibly a couple of volts either side of those given. Voltages at V+ and V– should be within about 250mV of the optimal values, while for DC supplies, the voltage at AGND should be almost exactly half that at V+.

If you've fitted sockets, cut power and plug in the remaining ICs. Don't get IC3 (TL072) mixed up with the other two ICs, which are LM833s. In each case, the pin 1 dot must go towards the top edge of the PCB, as shown in Fig.8. Re-apply power for the remaining steps.

Measure the voltage between the two test points labelled 'OFFSET' in the upper-right corner of the PCB. You should get a reading below 100mV. If not, switch off and check for soldering problems or incorrect components around IC3a and IC3b. Assuming the reading is low, slowly rotate trimpot VR3 and check that you can adjust it near zero. It should be possible to get the reading well under 1mV.

If you have appropriate cables or adaptors, you can now do a live signal test. Use a stereo RCA/RCA or



This is the 'DC' powered version of the *Spring Reverberation Unit*, as shown in Fig.8(a) above. The AC-powered version is slightly different, so if building that one, follow the overlay diagram shown above right.

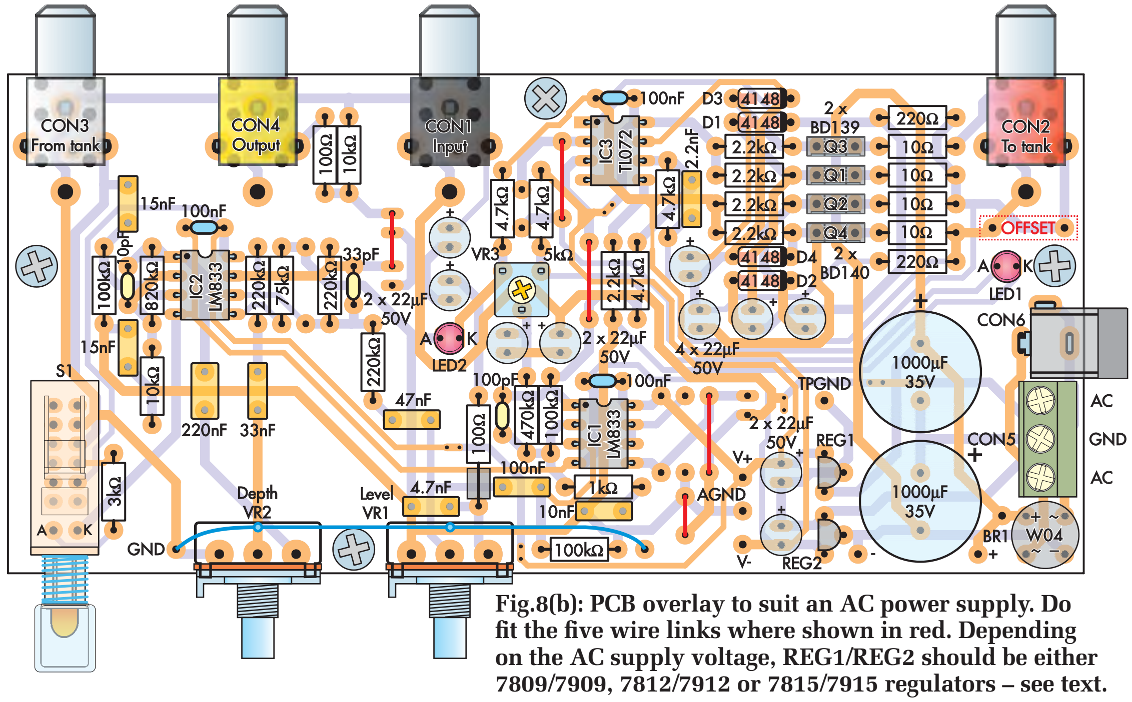


Fig.8(b): PCB overlay to suit an AC power supply. Do fit the five wire links where shown in red. Depending on the AC supply voltage, REG1/REG2 should be either 7809/7909, 7812/7912 or 7815/7915 regulators – see text.

RCA/3.5mm-plug cable to connect a mobile phone, MP3 player or other signal source to CON1. Turn VR1 and VR2 fully anti-clockwise. Use a cable with RCA plugs at one end and a 3.5mm stereo socket at the other end to connect a pair of headphones or earphones with a nominal impedance of at least 16Ω (ideally 32Ω or more) to CON4.

Power up the board, start the signal source and slowly advance VR1. You should hear the audio signal passing through the unit undistorted.

Now you can use a stereo RCA/RCA lead to connect the main board to the tank, via CON2 and CON3, matching up the labels on the board with those on the tank.

The tank should be placed on a level surface with the open part facing down. Continue listening to the signal source, then advance VR2. You should hear the reverb effect. If you're unsure, pause the audio source and you should continue to hear audio for several seconds until the reverb dies out.

That's it – the *Spring Reverberation Unit* is fully functional.

Installation

The tank should be installed with the open end down because the spring suspension is designed to work optimally in that position. Use the four corner holes to mount it since the tank is microphonic and these are designed to provide some isolation to prevent bumps from upsetting the springs too much. It would probably be a good idea to add extra rubber grommets under each spacer and avoid compressing them too much, for extra isolation.

As for mounting the PCB, you have three options. Option one is to mount it somewhere on the front panel of the amplifier so that switch S1 and potentiometers VR1 and VR2 are easily accessible. You then simply connect it to the tank using a stereo RCA/RCA lead.

If the panel it's mounted on is thin enough, it can be held in place using the two potentiometer nuts, although it would be a good idea to attach a small right-angle bracket to the mounting hole between the two pots, on the underside of the board via an insulating spacer, to provide a third anchor point on the panel.

The second possibility is to fashion a bracket from a sheet of aluminium with four holes drilled in it, matching the mounting holes in the board, with the side near the front of the board bent down and additional holes drilled in this flange for attachment to the front panel of the amp. You can then use self-tapping or machine screws to attach this bracket to the amp and then the board to the bracket. For bonus points, earth the aluminium bracket back to the GND pad on the PCB, to provide some shielding.

The third possibility is to leave S1, VR1 and VR2 off the board and mount it on top of the tank itself. We suggest using a long insulating spacer attached to one of the free holes on the tank's flange, supporting the PCB via the front or rear mounting hole, with a liberal application of thick double-sided foam tape on top of the tank to support the PCB.

You will need to trim the component leads carefully to make sure they can't poke through the foam tape and short on the top of the tank. In fact, it would be a good idea to silicone a sheet of plastic on top of the tank before applying the tape to provide extra insulation.

You would then mount S1, VR1 and VR2 wherever suitable and connect them back to the board using twin-core shielded cable for VR1 and VR2 (with the shield to the left-mount [ground] pin in each case). For the connections to S1, use regular shielded cable with the shield wired to the pin connected to ground and the central conductor for the audio pin, and a section of ribbon cable for the LED connections.

Using it

Using the *Spring Reverberation Unit* is straightforward. Push S1 in to enable reverb and push it again so it pops out to disable reverb. When reverb is enabled, S1 will light.

Adjust VR1 to give a near-maximum output level without clipping and then tweak VR2 until you get the desired reverberation effect.

With VR2 fully clockwise, the effect is overwhelming; you will probably find it most useful somewhere between 10 o'clock and 2 o'clock.

There are no controls on the spring reverb tank itself, just input (red) and output (white) RCA socket. All controls are on the PCB for this project.