CRAIG ANDERTON
ELECTRONIC GUITAR
Pedal Flanger, Part I

FOR MUCH OF 1979, THIS COLUMN focused on material related to pedalboards, switching tricks, and multiple-effects systems. Now it's time to get back into constructing some new effects, and we'll start off the '80s with a flanger.

Many of you have asked why I haven't covered a flanger in the past. Aside from complexity, the biggest problem was limited availability of Reticon's [910 Benecia Ave., Sunnyvale, CA 94086] SAD-1024 analog delay IC, the heart of the flanger. Now that this part is available from at least four or five different sources (which I'll mention later), we're ready to go.

First, we need to investigate how analog delay-line systems work, since flangers are analog delays. Actually, analog delays are very versatile systems: With short delays you get flanging; with slightly longer delays you get doubling; and with even longer delays you get echo effects. The principle of operation is so radically different from other effects devices that any time spent learning how they work will make construction, troubleshooting, and applying the flanger just that much easier.

Basic analog-delay circuitry. Fig. 1 shows a block diagram of a typical analog-delay system. If the delay time is short (from a few microseconds to five milliseconds or so), we'll hear flanging effects. A 10- to 30- millisecond delay causes a doubling or thickening effect, while a delay that's greater than about 50 milliseconds appears as echo. There are a number of delay-line ICs on the market; unfortunately, only a few of them are available to hobbyists for experimentation. The SAD-1024 is intended for short delays, which makes it ideal for flanging, while other chips like the Matsushita/Panasonic MN 3005 are more suitable for longer delays and echo effects.

Before considering the peripheral circuitry needed to support the delay line, let's consider how the delay line itself works. It still amazes me that an electronic circuit can actually sit there and produce time delays; however, by analyzing the delay-line circuitry we'll be able to de-mystify the process.

How delay lines work. Fig. 2 shows a simplified block diagram of an analog-delay chip with eight stages of delay (in practice, delay chips can have anywhere from dozens to thousands of delay stages). The clock shown in Fig. 1 is a timing reference whose usage will become clear as we go along.

The clock operates at a very high frequency, usually above the range of human hearing. Like most clocks, this one goes "tick-tock"; however, these are not audible clicks. Instead, they are voltages that swing between a maximum and a minimum value.

During the "tick" of the clock, the delay line "samples" the input waveform. Sampling is a process whereby the analog delay line stores the instantaneous amplitude value (level) of an output signal (see Fig. 3). If the signal at the moment of the sample is 1.1 volts, then the delay line stores 1.1 volts in stage 1. During the "tock" of the clock, the delay line stops sampling the input and transfers the voltage held in the first stage over to stage 2.

When the next "tick" happens, the delay line takes a second sample of the input signal and stores that sample in the first stage. During the "tock," this sample moves into the second stage, and the 1.1 volts we had stored in the second stage moves along to the third stage. As this process of moving samples down the delay line continues, you can see what happens: Eventually, the first sample will be shifted down the line far enough so that it appears at the output of the delay line, followed by the second sample, the third sample, and so on. So we have reconstructed our signal, but it is now delayed in time. In essence, we've broken the input signal down into a number of sampled voltages and passed these sampled voltages down the delay line.

Disadvantages of analog delay. This all looks good on paper, but there are some problems. First of all, in order to accurately represent the input signal, we need to take *lots* of samples--maybe one every 50 microseconds or so. This is like the connect-the-dots games that kids play; the more dots, the greater the resolution of the drawing. Our delay line's resolution works in much the same manner. However, as we increase the sample rate, we're also shifting our samples down the delay line at a faster rate, which gives us less delay. So, while we can get a pretty accurate output signal with short delay times, at longer delay times our sampling rate goes down and the signal becomes increasingly distorted.

One solution to this problem is to increase the number of stages, so that we can maintain a fast sampling rate but still get reasonably long delays. However, there is a flaw in this reasoning since each stage of delay creates a finite amount of noise, transfer inefficiency, and high-frequency loss which degrades the audio quality of the delayed signal. We therefore need to choose enough stages to give us the delay we want, consistent with a high enough sampling rate to give us good fidelity when we reconstruct our signal at the output of the delay line. If the clock goes low enough in frequency so that it's in the audio range, we now have an additional problem since we also hear the clock signal mixed in with the audio signal.

In addition, we have to figure out a way to change the delay. But this isn't too difficult: By making the clock frequency variable, we can change the overall delay by changing the clock frequency. As we slow down the rate at which the signal gets transferred from stage to stage down the line, the signal will take longer to appear at the output.

Peripheral circuitry. To minimize the problems associated with analog-delay lines, we can add the additional circuitry shown in Fig. 1. The input low-pass filter is not that critical; it's designed to keep high-frequency audio input signals from modulating the clock frequency. The output low-pass filter, however, is far more important: It removes the high-frequency clock signal from the audio, theoretically leaving us with the audio signal *only* at the output of the delay line. This filter also helps smooth the sampled signal into something that more closely resembles a "real" audio signal (see Fig. 3).

If we vary the clock frequency and listen to the output of the delay line, we'll hear pitch shifts and vibrato effects. In order to get flanging sounds, we need to mix the non-delayed signal and delayed signal together in exactly the same proportion. This causes cancellations and peaks in the response that give us the characteristic flanging sound.

Design tradeoffs. By keeping the clock speed fairly fast to make an adequate sampling of the input signal, and by being careful with our design, an inexpensive analog delay will still give us a reasonably good flanging effect. However, you should not expect miracles or performance rivalling expensive manufactured units—I think this column has probably shown just how complex the process of creating a time delay can be. You will get some noise, and you may hear the clock feeding through into the audio signal at long delay times. In addition, the high-frequency response will be limited by the response of the output low-pass filter. The unit we'll be building can be improved through the addition of extra peripheral circuitry, which we'll describe later on in this series.

In the meantime, start looking for the SAD-1024. This part is carried by Radio Shack, Adva Electronics [Box 4181, Woodside, CA 94062], and E-Systems [Box 5305, Berkeley, CA 94705]; PAIA [Box 14359, Oklahoma City, OK 73114] also offers an experimenter's kit that includes the SAD-1024 with a matching circuit board, so finding this part shouldn't present a problem. See you next month with a schematic.

Fig. 1.

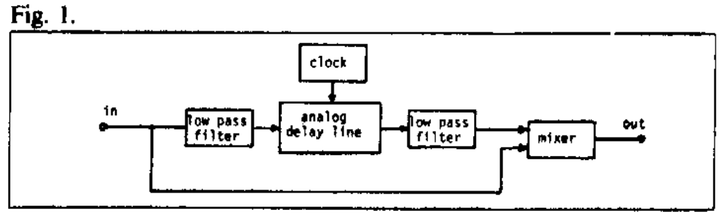
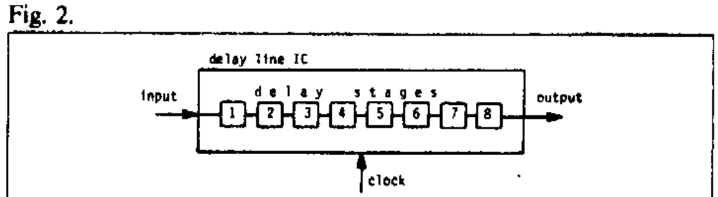
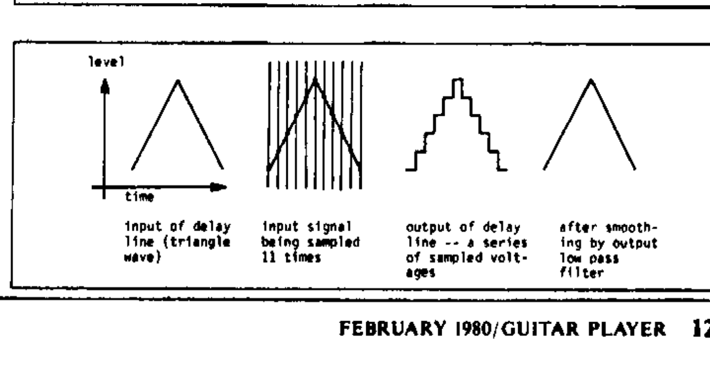


Fig. 2.





CRAIG ANDERTON
ELECTRONIC GUITAR
Pedal Flanger, Part II

THIS MONTH WE present the majority of the flanger circuitry (we'll show the clock circuit that connects to points A and B next month) and the parts list. This flanger started out as three pages of schematics and needed a well-equipped test bench for calibration; I realized that was a bit much to put in a mo th' column, so I published that design in iss 1:9 of *Device* [12304 Scribe Dr., Austin, TX 78759].

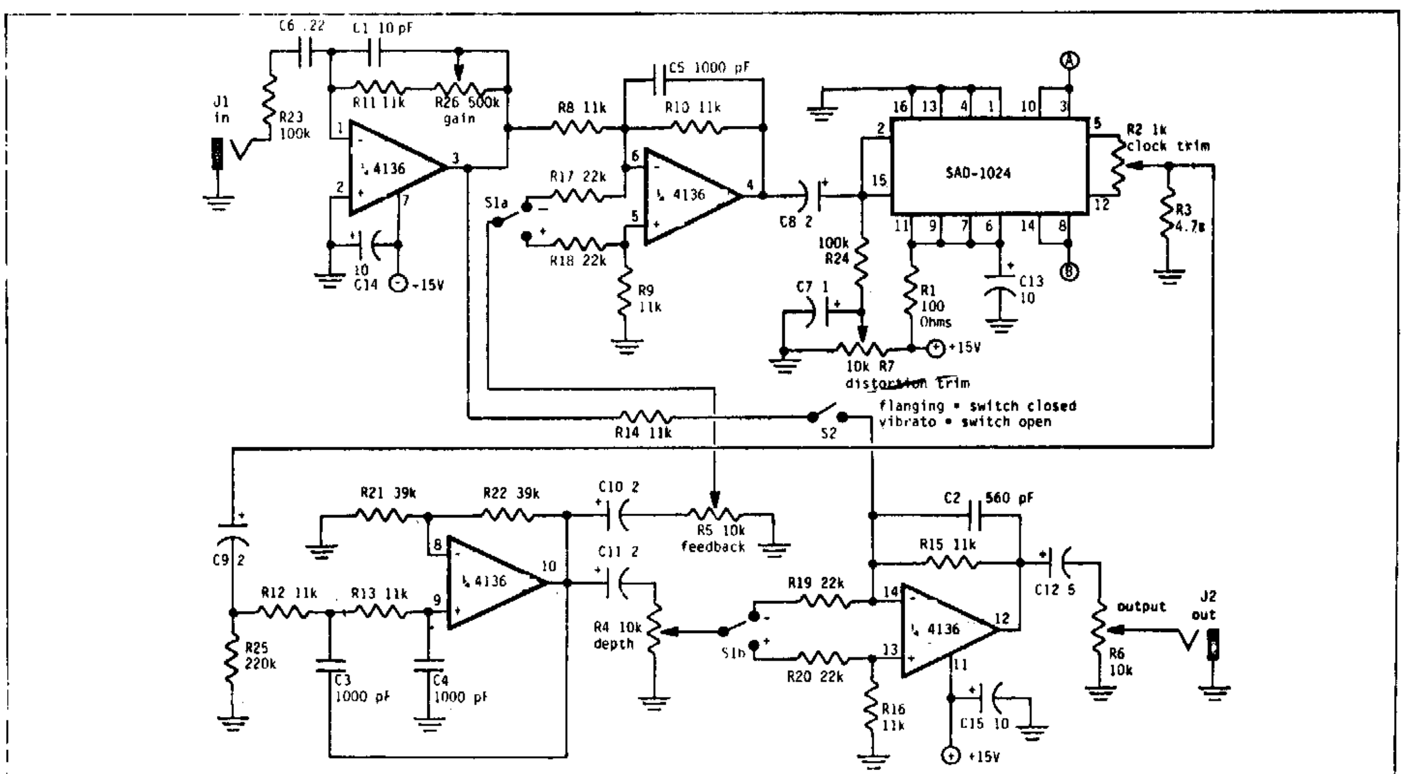
This new design is a considerably simplified version, yet it still offers excellent performance: The noise level is quite good (even without companding), there is a choice of positive or negative flanging sounds, it includes variable flanging depth and vibrato option, and, best of all, you can create a working flanger with a total of only three ICs. (You can't get much simpler than that and still have a unit with decent performance.) By the way, this unit is specifically designed to accept low-level inputs, but by changing one resistor, line levels can be accommodated.

Finding parts. I'm sorry to say there is no parts kit available for this project. However, we mentioned parts sources for the SAD-1024 last issue (Radio Shack stores for one), and the other parts are quite common. You'll note that I specified 11k resistors instead of 10k types; that's because 11k is a 5% value, and this degree of precision required for the best flanging sound. You can use 10k types in place of the 11k resistors if you check a batch of them with an ohmmeter and sort out the ones that are within 5% of each other.

Level of difficulty. This is not a simple circuit, but it's not impossible to build, either. You'll need to build an AC supply since batteries aren't really suitable, and you'll need to perform some careful calibrations to get the best possible performance. But if you're patient and follow the precautions given next month, everything should turn out all right. See you in April.

Flanger parts list

All resistors are ¼W, 5% (preferred)		C8 - C11	2μF, electrolytic or tantalum
R1	100 ohms	C12	5μF, electrolytic or tantalum
R2	1k trim pot	C13 - C15	10μF, electrolytic or tantalum
R3	4.7k	Semiconductors	
R4 - R6	10k audio or linear taper pot	IC1	RC4136 or XR4136 quad op-amp (Raytheon or Exar)
R7	10k trim pot	IC2	SAD-1024 serial analog-delay line (Reticon)
R8 - R16	11k	Mechanical parts	
R17 - R20	22k	J1, J2	Mono open circuit ¼" phone jacks
R21, R22	39k	S1	DPDT toggle switch (controls feedback phase)
R23, R24	100k	S2	SPST toggle switch
R25	220k	Miscellaneous parts: one 16-pin socket, one 14-pin socket, knobs, wire, solder, etc.	
R26	500k linear taper pot		
All capacitors rated 15 or more working volts			
C1	10pF ceramic disc		
C2	560pF ceramic disc		
C3 - C5	1000pF (polystyrene preferred, disc acceptable)		
C6	.22μF disc or mylar		
C7	1μF, electrolytic or tantalum		



CRAIG ANDERTON
ELECTRONIC GUITAR
Pedal Flanger, Part III

LAST MONTH WE PRESENTED the delay and audio sections for a good sounding, but simple-to-build, flanger. This month, we'll finish our discussion.

Clock circuitry. If the function of the clock is not clear, refer back to part 1 of this series, which appeared in the February '80 issue of *GP*. Fig. 1 shows the basic clock, based on a CMOS part called the CD4047. Points A and B connect to points A and B on last month's schematic; these wires should be as *short and direct as possible*. Points C and D can connect to one of the two auxiliary circuits shown in Figs. 2 and 3. Fig. 2 allows you to vary the flanging effect with a single pot; this is the best circuit to use when the pot can be mounted close to the board (say, within 5"). However, this option is most useful in studio and PA applications since it does not allow for easy footpedal control.

For footpedal or remote control applications, Fig. 3 represents a better option. The CD-4007 acts like a voltage-controlled resistor, with the voltage fed into pin 10 varying the flanging effect. Although the 4007 should be mounted close to the 4047, the pot may be mounted remotely from the rest of the 4007 circuit without degrading the circuit's performance. Some component juggling might be necessary for optimum performance; if the flanger effect doesn't go high enough in frequency, lower the value of the 27k resistor. If the flanging effect doesn't go low enough, lower the value of the 47K resistor. Since these two parts interact, if you change one you may have to change the other. But in most cases, these resistors will not have to be changed.

Power supply. For best results, the power for the flanger should come from a well-regulated, ±15 volt DC bipolar supply. Suitable kits are available from Bill Godbout Electronics [Bldg. 725, Oakland Airport, CA 94614] and from PAIA Electronics [1020 W. Wilshire Blvd., Oklahoma City, OK 73116 (stock #4771)]. Also, Bernie Hutchins has published several suitable designs for do-it-yourselfers in *Electronotes* [1 Pheasant Lane, Ithaca, NY 14850].

Construction tips. Here are several tips, in no particular order:

1. The 47pF capacitor in the clock circuit (Fig. 1) should be a stable polystyrene type for minimum flanging variations with changes in temperature.

2. The clock circuitry creates very high-amplitude, high-frequency signals. They can be picked up by sensitive electronic circuits (such as preamps and fuzztones), and could possibly cause problems by coupling into these stages, or into the preamp or mixer stages of the flanger. Therefore, use shielded cable to connect J1 to the input of the flanger, and also run shielded cable between S2 and pin 14 of IC1. Actually, your best approach is to carefully consider the circuit board layout before you start wiring: The clock circuitry should be mounted as close as possible to the delay line, but away from the audio stages based around IC1 as well as the various switches and controls.

While the layout isn't critical, any extra care spent in layout will pay off dividends in improved performance. Avoid long power and ground lines; use a single-point grounding system as much as possible; and keep all audio lines as short as possible. If you are powering other modules along with the flanger, connect a 10uF to 100uF capacitor (at 15 working volts DC) across the power supply lines connecting to the 4047 (the plus end of the capacitor connects to pin 14, while the minus end connects to pin 7). Again, this may not be necessary—but it certainly can't hurt.

3. IC handling: The clock ICs and delay-line IC may be destroyed by improper handling due to their sensitivity to static electricity charges. For best results, and so you don't have to kiss an $11 chip goodbye, *use sockets for these ICs* and keep them in their protective foam until just before it's time to test the circuit. Then, touch something grounded or metallic (to drain off any residual charge which you may have accumulated, say, by walking across a rug on a dry winter day) and pop the ICs into their sockets.

Calibration. Begin by opening S2 (vibrato position). This will allow you to hear only the delay section, thus simplifying calibration. Turn R4 up all the way; then turn R5, R6, and R26 down all the way. S1 can be in either position, and the 100k control used for the clock should be at the approximate halfway point.

Next, plug your instrument into J1, hook up J2 to your amp, turn on the power supply, and turn up R6 halfway. Play some loud chords with R26 turned up just a little bit; you may or may not hear anything. Now rotate R7 until you get the least distorted sound out of the flanger. Increase R26 until you get *just* distortion, then readjust R7 for the least distorted sound. Continue increasing R26 and readjusting R7 until you cannot get a clean sound anymore; then back off a bit on R26 until the sound cleans up. This is the optimum setting for R26. If you encounter distortion when playing, make sure R7 is properly calibrated; if it is, turn down R26 a bit further. If you have a scope, connect its probe to the midpoint of R2 and adjust this trimpot for the cleanest (least fuzzy) looking waveform. Otherwise, leave R2 at the halfway point.

Using the flanger. Close S2, turn up R4 all the way, adjust R6 to suit, and vary the 100k clock control: You should now hear that familiar flanging sound. Next, increase R5 until it's just short of the oscillation (squealing) point; this should dramatically increase the intensity of the flanging. Be forewarned—when the flanger goes into oscillation from turning R5 up too high, it can get *loud*; so keep your amp volume down a bit lower than normal as you experiment with different settings of R5. Change S1 to its other position; again, experiment with the setting of R5. In the (+) position of S1, the flanging sound is sharp and metallic, while the (-) position gives a more tubular and round sound.

Modifications:

1. If you want more output level, double the value of R15 and halve the value of C2.

2. If you want to use the flanger with line-level signals, change R26 to 100k (preferred method) or increase R23 to 470K (this is easier, but also contributes a little more noise).

3. To experiment with slapback echo effects, change C3 and C4 to .01uF Mylar capacitors and change the 47pF clock timing capacitor to 220pF. Don't expect any wonders in terms of performance, but you might enjoy using the flanger in this manner for some applications.

Final comments. In order to avoid taking up another installment of the column, I've tried to be as concise as possible. I hope this doesn't scare off too many builders; I have a lot of confidence in this design, am extremely happy with the way it sounds, and would like to make sure that anybody who wants to build it has enough information to do so. If there's anything you don't understand about this project, feel free to write me c/o *GP*, and we'll cover your queries in the next questions-oriented column.

Fig. 1.

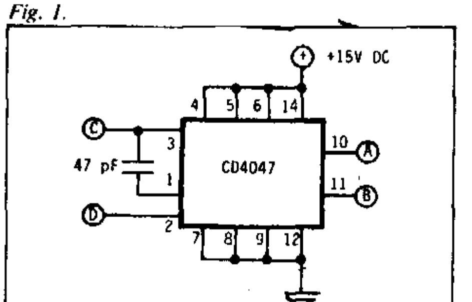


Fig. 2.

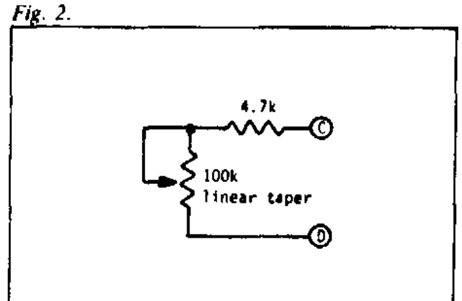
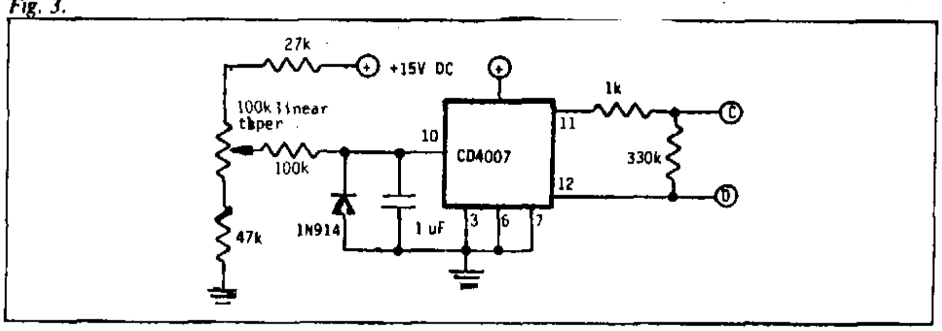
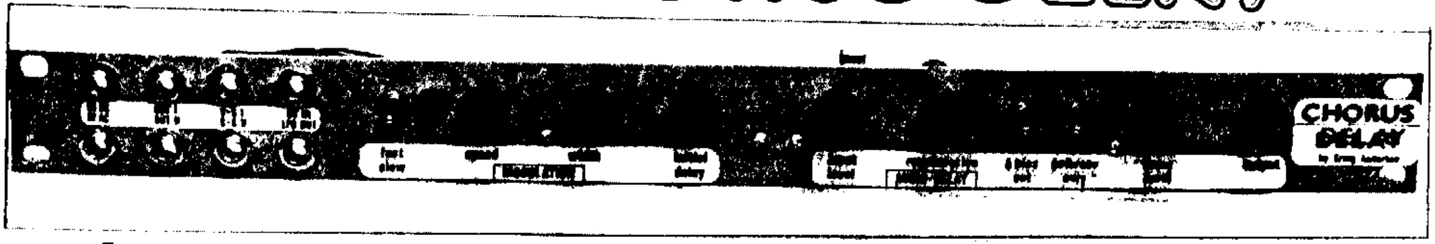


Fig. 3.



Build a CHORUS·DELAY



Chorus·Analog Delay·ADT·Flanger·Vibrato

By Craig Anderton

MANY *GUITAR PLAYER* readers have written to me requesting a chorus box project, but the device we'll examine does a lot more. By covering the range from 80 ms (milliseconds, or thousandths of a second) to 2 ms, it also delivers excellent flanging, slapback echo, and ADT (automatic double tracking) effects. Note, however, that this is not a simple circuit. I only recommend it for those of you who are proficient at building circuits from scratch. If you *do* have the requisite electronic chops, though, you will be rewarded with a wide-range delay line that is not only quiet, but also more flexible than many other chorus units.

This feature-length article is a departure from my usual single-page column format. However, there is still more to say about this device than can even fit in an article of this length. So, in next month's Electronic Guitarist column I will continue this feature by getting into some relatively esoteric modifications and customizing.

How It Works

The Chorus/Delay has two major submodules, the audio delay section (Fig. 1) and the modulation section (Fig. 2). The heart of the audio delay section is the Reticon SAD-4096 integrated circuit, a member of the bucket-brigade analog delay line family (for more information about what makes delay lines tick, refer to my column in the Feb. '80 issue of *Guitar Player*). This is an expensive chip—it lists for about $50.00—but it can give a 100ms (1/10 of a second) delay with excellent fidelity. An added advantage of the SAD-4096 is that it can be driven by a pulse-generating clock at very high speeds, giving delays as short as 2ms (which puts you right in the flanging range).

However, like all delay lines the SAD-4096 is inherently noisy and has poor high-frequency response. We attack the noise problem with IC2, the NE570/571 compander from Signetics. The first half, IC2A, is a *compressor*, which compresses the dynamic range of the signal going into the delay line, thereby maintaining a high average signal level through the SAD-4096 and keeping the signal out of the "noise floor" of the chip. The second half, IC2B, is an *expander*, which takes the delay line's output and restores the signal to its normal dynamic range; the expanding action also lowers the noise floor by around 25dB. The result is dead-quiet operation up to about 50ms, and only minimal noise out to the full 80ms. This companding type of noise reduction system is similar to the fancy noise reduction systems found in recording studios.

> **Caution**
>
> The Chorus/Delay is a complicated project with sensitive integrated circuits that are relatively expensive. Do not attempt to construct this circuit unless you have had previous experience building electronic effects.

To further reduce noise and improve the high-frequency response as well, resistor R3 and capacitor C7 boost the highs going into the compressor and delay. R6 and C6 cut the highs by a complementary amount, both to reduce any residual hiss and to compensate for the boost created by R3 and C7. The rest of the circuitry in Fig. 1 involves IC1D (a buffer), IC1A (which drives the delay line), IC1B (which filters out most of the high-frequency clock signal from the low-frequency audio signal), and IC1C (the mixer that combines delayed and normal sounds).

The modulation circuitry (Fig. 2) provides the high-frequency clock necessary to drive the SAD-4096, and an LFO (low-frequency oscillator) to modulate that clock. The SAD-4096 accepts a clock frequency from 8kHz to 1MHz. Our clock circuit runs from 25kHz to 1MHz to simplify the audio filtering requirements. IC1A and IC1B make up the LFO, while IC1D drives the two monitor LEDs, which are connected back-to-back. As the modulating triangle wave produced by the LFO goes more positive, D3 gets brighter; as the triangle wave goes less positive, D3 gets dimmer; then as the wave goes below ground (0 volts) and starts turning negative, D2 becomes brighter. When the wave starts turning positive again, D2 becomes dimmer until it extinguishes and D3 then starts getting brighter again. This process repeats for every cycle of the triangle wave. Watching these blinking lights is a great way to monitor an LFO. You could also use different colored LEDs for D2 and D3, or you may choose a two-color LED (which Radio Shack sells) for some even more amazing effects.

The remaining op-amp, IC1C, is a simple filter that turns the triangle wave into more of a sine wave at faster LFO speeds. It also mixes the initial delay and modulation control voltages.

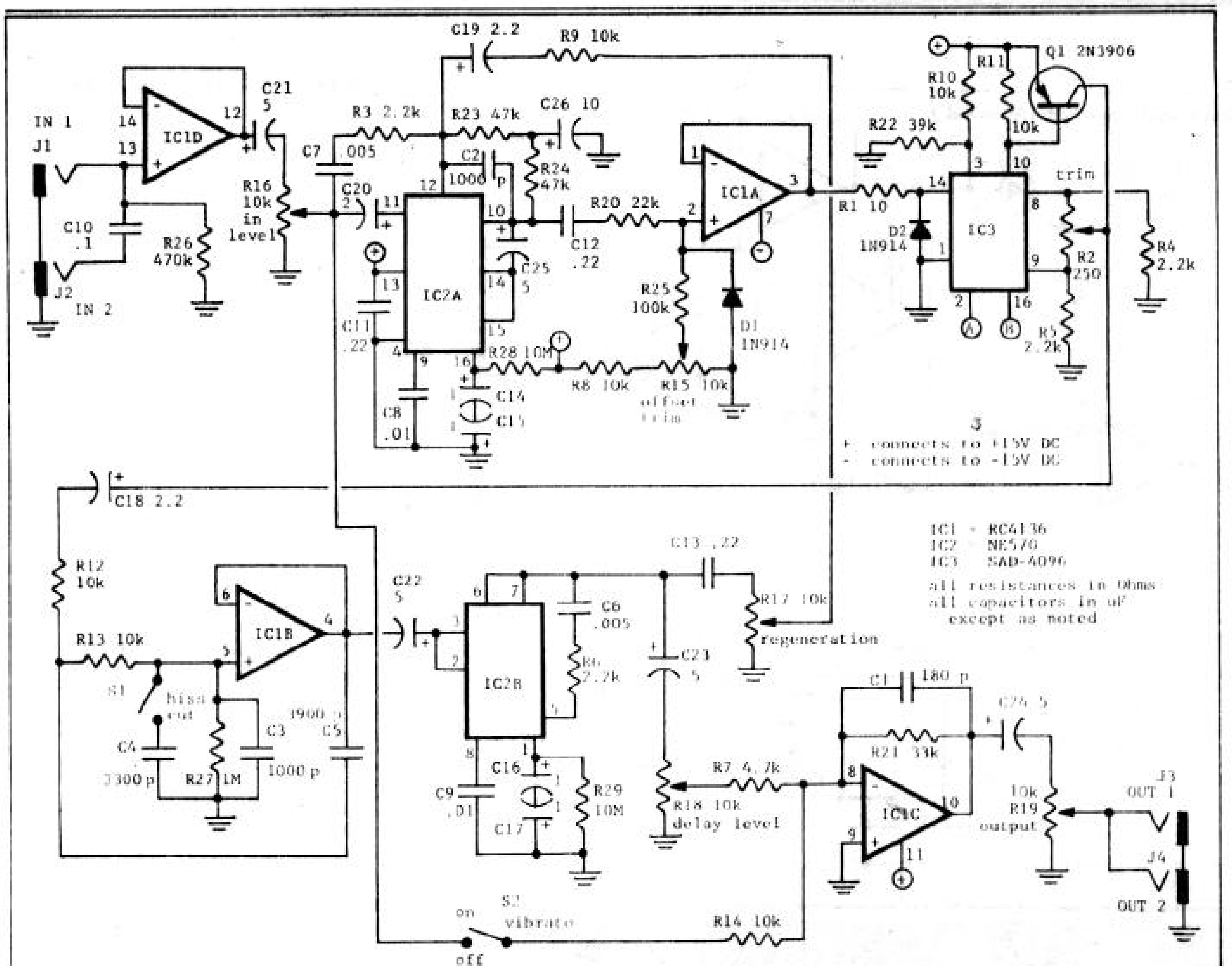
Circuit Board Construction Tips

Remember, this is a complicated circuit that uses radio, as well as audio, frequencies. You must pay attention to the following instructions, or you may end up with a unit that works poorly—if at all.

1. Use single-point grounding for each circuit board wherever possible. Ideally, you should return all grounds to a single point on the PC (printed circuit) board. Since this may not be practical, you should at least return the grounds for each stage (compressor, expander, delay, buffer, mixer, etc.) to one ground point. The ground points for each board should return separately to the

Craig Anderton's work has been published in numerous electronics magazines, including Device, Polyphony, *and* Modern Recording. *He is the author of* Electronic Projects For Musicians *and* Home Recording For Musicians [*Music Sales, 33 W. 60th St., New York, NY 10023*], *and a monthly* GP *column. He has also written construction articles on practice headphones* [*Jan. '75*] *and a treble booster* [*Apr. '75*], *and an overview of effects* [*May '76*]. *His Feb. '82 column will cover ways to use and modify his Chorus/Delay.*

PHOTOS BY JON SIEVERT



Audio Delay Board Parts List

Resistors (all 10%, 1/4 watt)

R1	10 ohms
R2	250 ohm trimpot [symmetry trim]
R3-R6	2.2k (2k2 metric)
R7	4.7k (4k7 metric)
R8-R14	10k
R15	10k trimpot [offset trim]
R16	10k pot, log taper preferred [input level]
R17	10k pot, linear taper preferred [regeneration]
R18	10k pot, log preferred [delay level]
R19	10k pot, log preferred [output level]
R20	22k
R21	33k
R22	39k
R23, R24	47k
R25	100k
R26	470k
R27	1 Meg
R28, R29	10 Meg

Capacitors (all rated at 15V or more)

C1	180 pF, polystyrene preferred
C2	1000 pF (1 nF metric), polystyrene preferred; see text
C3	1000 pF (1 nF metric), polystyrene preferred
C4	3300 pF (3n3 metric), polystyrene preferred
C5	3900 pF (3n9 metric), polystyrene preferred
C6, C7	.005 μF (5 nF metric), mylar preferred
C8, C9	.01 μF (10 nF metric), disc or mylar
C10	.1 μF (100 nF metric), disc or mylar
C11-C13	.22 μF (220 nF metric), mylar preferred
C14-C17	1 μF, tantalum preferred
C18-C20	2.2 μF (2u2 metric), tantalum or electrolytic
C21-C25	5 μF tantalum or electrolytic
C26	10 μF tantalum or electrolytic

Semiconductors

D1, D2	1N914 small-signal silicon diode
IC1	RC4136 or XR4136 quad op-amp
IC2	NE570 or NE571 compander
IC3	SAD-4096 analog delay line
Q1	2N3906 PNP transistor

Mechanical parts

S1	SPST switch [hiss cut]
S2	SPST switch [vibrato on/off]
J1-J4	Mono open-circuit 1/4" jack
Misc.	Knobs, solder, wire, sockets, power supply line bypass capacitors, etc.

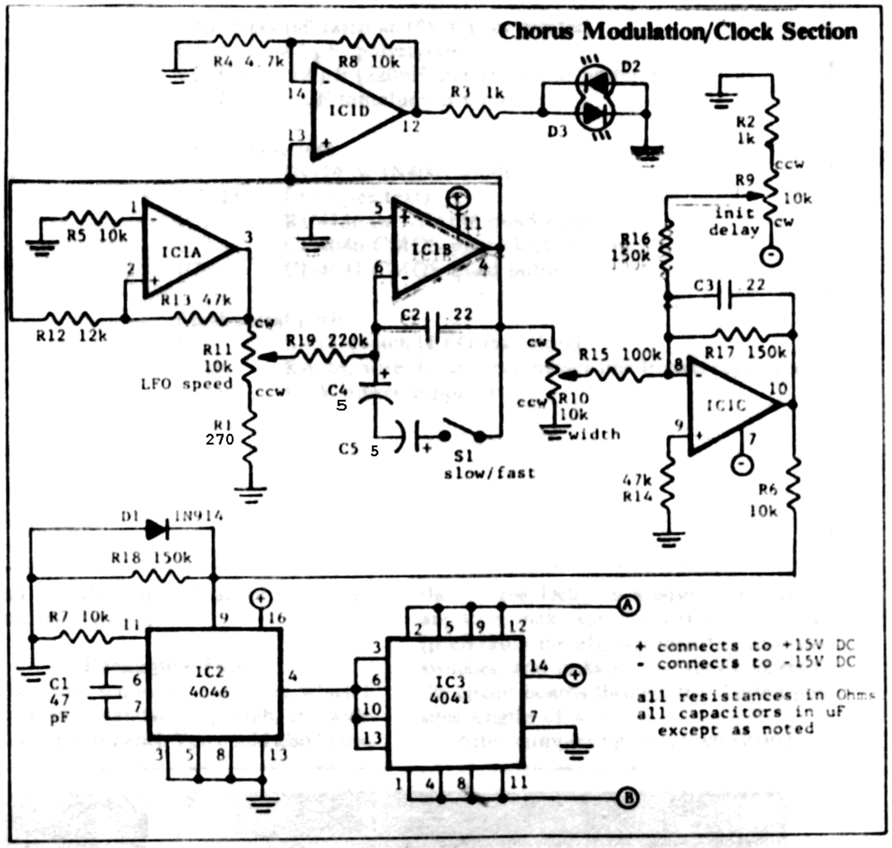
power supply or main chassis ground.

2. Keep leads as short and direct as possible.

3. Add the power supply bypassing shown in Fig. 3 where the power supply lines enter the board, and check on the audio delay board tht C11 is mounted as closely as possible to pins 13 and 4 of IC2A. It's also good practice to solder some .1 μF ceramic disc capacitors right from the power supply pins of the 4046, 4041, and SAD-4096 to ground. If you're making your own PC board, do not drill holes for these capacitors in the board; rather, solder them *directly* to the IC socket pins on the *foil side* of the board, keeping the capacitor leads as short as possible.

4. Arrange the boards so that pads A and B from the audio delay board can connect to pads A and B on the modulation board by the shortest possible length of wire. When you connect these points, use heavy-gauge wire to minimize inductance and resistance effects (the 4041 power buffer has to overcome the SAD-4096's clock line capacitance—about 1000 pF!). And while we're on the subject of wiring, use *shielded* cable on any audio lines that run for more than a few inches.

5. Power both boards with a well-regulated ±15 volt DC power supply, such as the kind used for keyboard synthesizers. These higher voltages are required for optimum operation of the Chorus/Delay.



Chorus Modulation/Clock Section

Chorus Unit Modulation/Clock Parts List

Resistors (all 10%, 1/4 watt)

R1	270 ohms
R2, R3	1k
R4	4.7k (4k7 metric)
R5-R8	10k
R9	10k pot, log preferred [initial delay]
R10	10k pot, log preferred [LFO width]
R11	10k pot, log preferred [LFO speed]
R12	12k
R13, R14	47k
R15	100k
R16-R18	150k
R19	220k

Capacitors (all rated at 15V DC or greater)

C1	47 pF, polystyrene
C2, C3	.22 μF (220nF metric), mylar preferred
C4, C5	5 μF tantalum or electrolytic

Semiconductors

D1	IN914 or IN4001 diode
D2, D3	LED (see text)
IC1	RC4136 or XR4136 quad op-amp
IC2	CD4046 CMOS phase-locked loop
IC3	CD4041 CMOS quad buffer

Mechancial parts

S1	SPST switch [LFO fast/slow]
Misc.	Knobs, wire. IC sockets, power line bypass capacitors, ±15V power supply, etc.

6. Take your time and work carefully. Impatience, more than ignorance, scuttles otherwise successful projects.

7. Use sockets for all ICs, and follow all specifications given in the parts list (if it says a *mylar* capacitor is preferred, do your best to find a mylar cap instead of a cheaper ceramic disc type).

Packaging Tips

The photo shows my prototype, which I mounted in a standard 1¾" high, 19" wide rack-mountable panel. You could also house the Chorus/Delay in a large floor box. In any case, make sure it's a sturdy enclosure (preferably metal). Arrange the controls, switches, and jacks so that they connect to the circuit boards through the shortest possible lengths of wires.

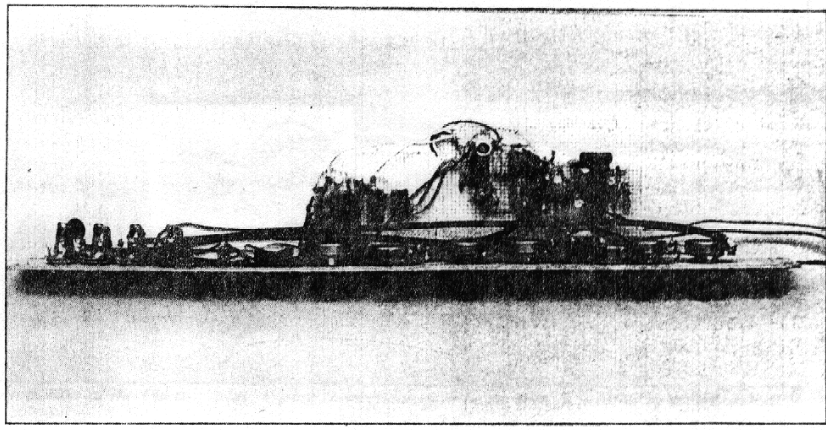
After completing circuit construction,

mount both PC boards to the front panel. Next, wire the board connections to the pots, switches, and jacks; connect points A and B from both boards; and finally, connect the ground and power supply wires. Check your wiring over carefully, and prepare for calibration by studying the next paragraph.

Calibration

Calibration is fairly simple, and only involves making two adjustments on the audio delay board. Set all controls as follows: LFO speed, LFO width, and regeneration completely counterclockwise; initial delay and output up halfway; input level and delay level fully clockwise; vibrato switch open (*on* position); hiss cut switch open; and fast/slow switch open. Plug your guitar into the input; J1 is the input of choice, since it direct-couples your signal into IC1D. If you experience problems using this input, plug into J2 instead. Chances are, you will either hear no sound at all or a highly distorted sound. Take heart; relief is only a trimpot away.

Start varying offset trimpot R15. At one position in its rotation, the distortion will disappear; as you move away in either direction from this magic space, the distortion will return. If distortion happens at all settings, then set the trimpot where distortion is least, trim back a bit on the input level control, and try again. The SAD-4096 can only accept



Top view of the Chorus/Delay shows the clock/modulation board (L) and the signal processing board. All switches, pots, and jacks are mounted to a single 19"-wide panel.

about a 2-volt peak-to-peak signal, as the input level allows you to reduce the level of strong signals going to the delay line.

There are two ways to calibrate trimpot R2. The preferred method is to monitor the middle terminal of this trimpot with an oscilloscope; you will observe two thick lines (possibly along with some thinner lines) when you're not playing. Rotate R2 until the two thick lines converge into one. If you don't have an oscilloscope, short out R2 on the modulation board with a jumper wire that has alligator clips on both ends. Now, adjust the initial delay control until you hear a very high frequency whine (the highest you can hear), then adjust R2 for the *minimum* amount of this whine.

Learning The Controls

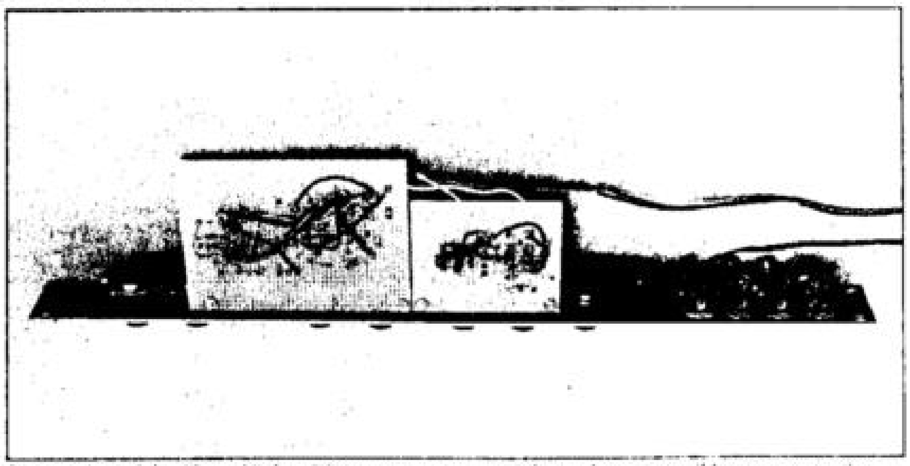
First, a warning: Unlike most chorus boxes, which are optimized for the chorusing effect and therefore have very few controls, the Chorus/Delay has lots of controls. This gives you the capability of making some very beautiful sounds, but it also gives the freedom to make some really ugly ones as well. For example, if you use as much modulation for chorusing as you do for flanging, the

sound will resemble banshees in heat; conversely, if you use as little modulation for flanging as you do for chorusing, the flanging effect will be barely noticeable. So, be patient not only with respect to constructing the Chorus/Delay, but also during the checkout as well.

Begin by setting all controls and switches as mentioned in the section entitled "Calibration." Now, close the vibrato switch (vibrato "off") and play. If you hear distortion, either back off on the input level control or check that the offset trimpot is properly calibrated. Next, adjust the output control for a pleasing level and you're ready to go.

Vary the initial delay control. As you turn the control clockwise, you get into the flanging range; as you turn it counterclockwise, you start hearing echo effects. At extreme counterclockwise positions, you may notice some hiss. In this case, close the hiss cut switch. However, for maximum high-frequency responses it's best to leave this switch open unless the hiss bothers you.

Next, try advancing the regeneration control; it governs the amount of signal feedback. In the flanging range, this will give a more metallic sound. In the echo range, you'll hear multiple echo repeats. Note also that in the echo range you'll probably want to turn the delay level control down a bit for a more realistic effect—after all, natural



Bottom view of the Chorus/Delay: Wiring connections must be as short as possible to prevent noise.

echoes aren't as loud as the primary sound source.

Now it's time to experiment with modulation. You will want to add the deepest modulation in the flanging range. Turn the width control up full; then adjust the initial delay clockwise for the widest sweep range. You may not want this much modulation at all times, but it's nice to know it's there when you need it. In the echo and chorusing ranges, even moderate amounts of modulation will add grotesque pitch-bending, so you may want to keep this control subdued.

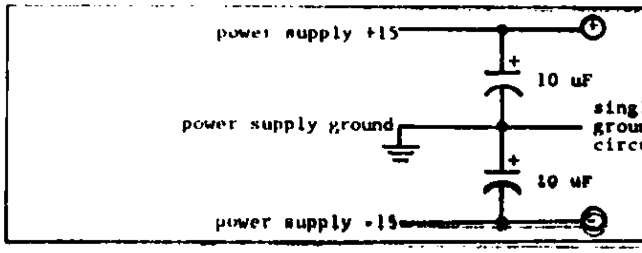
I find my favorite chorus setting by picking some muted strings on the guitar to produce the sharpest possible sound. I then set the initial delay control so that there is a distinct, short echo just starting to appear. I then return the initial delay clockwise a bit until the two echoes barely fuse into one. Finally I add modulation to suit. A little regeneration adds a nice touch for a spacier, more sci-fi chorus sound.

From this point on, you're on your own. Just remember that it will probably take you a lot of time to learn what all these controls do and how they interact. This is very flexible unit.

Finding Parts

I've been itching to present some delay line projects for a couple of years now, but could not locate a reliable and inexpensive source for either the Panasonic delay line chips or the long-delay Reticon chips (SAD-1024s are easy to find, but they aren't suitable for echo). Luckily, PGS Electronics [Box 735, Terre Haute, IN 47808] now carries the SAD-4096 for $31.00 plus $1.50 postage and handling. At my request, they also offer a package of all the IC's needed to complete this project (4046, 4041, 570, 4096, and two 4136s) at a discounted price to *Guitar Player* readers of $36.00 plus $1.50 postage and handling; foreign orders add $1.00 extra for air mail postage. PGS also has pre-etched printed circuit boards ($4.50 for the modulation section; $6.50 for the delay section; $2.50 for positive or negative artwork. Add $1.50 per order for postage).

All other parts are commonly available through any of dozens of mail-order electronics outlets; most parts for this project are also available through Radio Shack. In addition, check the ads in the back of *Radio Electronics* and *Popular Electronics* magazines to find more parts sources. **Note:** Please, don't write me and say you can't find the parts anywhere — I've given you a source for the semiconductors and the other parts are not at all hard to find.



Chorus/Delay Power Supply Filtering

Conclusion

There's much more that could be said about the Chorus/Delay — how to apply it, studio tricks, and so on. Here's one example: On the audio delay board there's a C2, a 1000pF capacitor that restricts the high-frequency response of the compressor. This is designed to filter out extreme high frequencies going into the delay line, but in practice you might not need it at all, or you may be able to get by with a lower value (such as 220pF), the lower the value, the better the high-frequency response. I wanted a little more zing in the sound — more highs and brightness — so I removed this capacitor; the Chorus/Delay still works just fine with instruments such as the guitar, that don't have outrageous amounts of high frequencies.

There are other ways to customize the performance of the chorus, and we'll discuss them in my regular column next month. You might also notice that the prototype in the photo has eight jacks instead of the four given in the schematic; next month I'll explain what they're all about as well.

So far I've stressed that this circuit is complex. In a spirit of equal time, here's some encouragement: After prototyping the circuit on a breadboard, I drew up a schematic and built the unit in the photograph. Normally, you might expect that I would have had to make a few adjustments or changes in the transition from breadboard to real circuit; fortunately, the whole circuit went together flawlessly and gave me no trouble whatsoever. I hope it's as trouble-free for you!

Acknowledgements

I'd like to thank Dave Tarnowski of A/DA for procuring an SAD-4096 for me back when no one else could, and for suggesting the use of the 4041 as a high-current driver for this chip. I'd also like to thank Greg Schneck at PGS Electronics for making the SAD-4096s and PC boards readily available.

CRAIG ANDERTON

ELECTRONIC GUITAR

Chorus/Delay Mods For Even More Sounds

BY NOW YOU PROBABLY have last month's Chorus/Delay unit perking along and giving you all kinds of neat effects. The next step might be to add some modifications that personalize the unit; then again, you may be more experimentally minded, and wondering just what other games you can play with the device. This column discusses these options, as well as explains a little more about how the delay works. Bear in mind, though, that this discussion is intended for *more advanced* experimenters.

Modulation section modifications. The first modification involves two extra jacks. Break the connection going between the width control (R10) and pin 4 of IC1, as shown in Fig. 1. Next, insert two jacks, one closed-circuit and one open-circuit.

The LFO (low-frequency oscillator) output jack taps off the LFO output. Thus, if you build two unit and wish to drive them from the same LFO, you can patch the LFO output from one unit to the CV input on the other unit. This will sync the two together.

The CV (control voltage) input accepts external control voltages, like the kind generated by synthesizers. Also, some of the more sophisticated effects have control-voltage outputs. For example, some envelope followers include an envelope output: You can patch this into the delay's CV input, and have the chorus track your dynamics. This is a great effect, and imparts what I can only describe as a "scintillating" sound to your instrument. In addition, you can trim the overall response of this input with the width control.

Another very powerful use of these jacks is with a footpedal. Patch the LFO-out jack to the pedal input, and the pedal output to the CV-in jack. This gives you foot control over the degree of modulation. While this is useful for chorus and flanging effects, it is perhaps most useful with vibrato. Pulling back on the pedal gives no vibrato, while pushing down all the way gives a throbbing vibrato. In between, you can vary the vibrato depth from subtle to dramatic. Remember, too, that unlike finger vibrato—where you can only vibrato one or two strings at a time—electronic vibrato works on the entire instrument. Chords with vibrato are something else altogether! Also, using this effect with vocalists who don't have natural vibrato can help the character of the voice tremendously.

The two other jacks I've added to the delay are for *synchro-sonic* applications. Without getting too heavily into the theory behind synchro-sonics, the basic premise is that in a fully synchro-sonic system, all rhythmic events—including vibrato, tremolo, synthesizer envelopes, LFOs, etc.—may be synchronized (if desired) and traced to one master clock. By adding the circuit shown in Fig. 2, between pins 4 and 6 of IC1B, a reset pulse at the synchro-sonic input forces an LFO reset, and gives retriggering. Should this retriggering occur too abruptly for your tastes, add a low-value resistor (100 ohms to 1k) in series with the collector of the transistor. Now I know all this synchronization talk will mean absolutely nothing to many readers, but the few who are into this type of thing should find the synchro-sonic input most valuable. Incidentally, the reason for using two jacks is so that a synchro-sonic pulse can be routed both to and from the delay module.

There's one more modulation section modification, and that's increasing the delay time. Unfortunately, at these long delay times the sound is going to be noisy, distorted, and generally rotten; so, don't say I didn't warn you! But if you're willing to greatly decrease the high-frequency response (we'll cover that later in this column), the longer delay time can be useful. Simply decrease R2 to, say, 470 ohms for starters. The lower the value of this resistor, the longer the delay you'll be able to get. Once the resistor gets too low, though, the clock sound will become unbearable and eventually the delay line will just poop out altogether. You can also shift the delay time lower by increasing C1 to 100 pF.

Audio delay section modifications. If you experiment with longer delay times, you'll have to reduce the bandwidth of the output smoothing filter. This involves increasing the values of C3 and C5. C5 should be four times larger than C3; for example, if C3 = .005 μF, then C5 = .02 μF. For a quick and dirty way to cut high-frequency response, just add a big capacitor in parallel with C3 (try .01 μF). If you want to add echo to a bass drum, this will do the job.

The response of the regeneration control has been altered by C13, which rolls off the lower frequencies. This prevents boomy, bassy "howls" and peaks that can occur with extreme amounts of resonance. However, if you feel like taking out some aggressions on a nearby woofer, you can increase C13 to 1 μF and give the regeneration circuit full bass bandwidth.

Another set of modifications involves the pre-emphasis/de-emphasis circuit. R3/C7 boost the treble going into IC2A; R6/C6 cut the treble by a complementary amount coming out of IC2B. However, the delay line contributes some high-frequency rolloff (whether you want it or not), as does the output filter. If you want to increase the pre-emphasis, decrease R3 to 1k. This will give you a brighter sound. Or, if these extra high frequencies cause problems (such as aliasing—glitches resulting from generated frequencies too high to be discreetly sampled by the unit), you can cut the value of C6 instead to .002 μF or less. This will create less of a treble-cutting effect; the tradeoff is that the noise will appear more prominent.

Some instruments, such as guitar synthesizers, have a great deal of high-frequency response, these might interact with the delay line clock to produce undesirable non-harmonically related sounds in the output. Should you experience this problem, remove the pre-emphasis and de-emphasis components. The noise will be more apparent, but I think that a little extra noise is preferable to aliasing sounds. In extreme cases, you can also increase the value of C2.

Finally, if you want a little more output level, you can increase R21 to 47k, or even to 100k. On the other hand, if the chorus is too hot and overloads your guitar amp, you can reduce R21 to a lower value (such as 10k).

So much for the Chorus/Delay unit; I hope you enjoy using (as well as modifying) it. I'd be interested in any of your experiences: Did it go together easily? Would you like more complex projects like this? Write to me c/o *Guitar Player*.

Next month, we'll get back to basics—part one of how to read schematics. See you then.

Fig. 1.

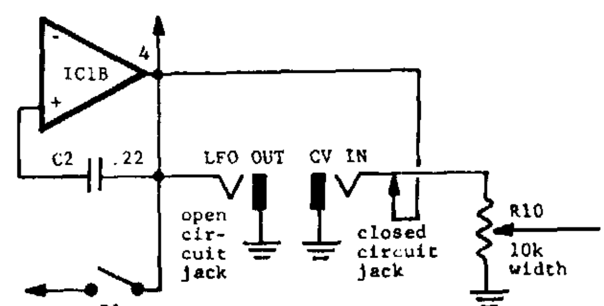


Fig. 2.