MIDWEST ANALOG PRODUCTS

Kits, Parts and Plans for the Do-It-Yourself Electronic Musician

Voltage Controlled Analog Delay

Private Collection — Information Packet

Part #K140

ABOUT THE PRIVATE COLLECTION

The *Private Collection* is a new series of special offerings from Midwest Analog Products, available in limited quantities only. As the name implies, this is a line of items for the connoisseur of analog synthesis—for those of you with a more advanced background. If you know how to read schematics, etch circuit boards (using the artwork we provide) and have the shop skills to fabricate your own rack panels, then here is your chance to save money while building some of the newest and most exciting designs ever from Thomas Henry.

In this exclusive line of kits, we include only the hard-to-find components (you can probably save money by buying the common parts from any of a number of mail order houses, or maybe you already have them in your own lab). We also provide high quality 1:1 printed circuit board artwork that you can use to etch your own board at home. (We recommend PNP Blue "Press-n-Peel," made by Techniks, Inc. Just photocopy our artwork onto a PNP Blue sheet, iron it onto a blank board and start etching!)

Also included is the hardcore info you need to start building at once: things like rack panel layouts, drilling guides, decal artwork you can photocopy onto clear sticky-back sheets, and so on. And of course, each kit includes complete schematics, circuit description and construction tips.

So, if you're a more advanced electronic musician looking to outfit your studio with high quality gear no one else has, then check out our entire *Private Collection* series!

WE WANT YOU TO BE SATISFIED WITH YOUR PURCHASE

If after looking over this *Information Packet* you feel that the **Voltage Controlled Analog Delay** is beyond your abilities, return the unopened bag of parts, this *Packet* and all other printed sheets for a full refund. See the Return Policy on page 10 for details.

ABOUT THE VOLTAGE CONTROLLED ANALOG DELAY

This is our first entry in the *Private Collection* series: a **Voltage Controlled Analog Delay** module featuring the super wide range SAD-4096 bucket brigade chip. This circuit (along with most any LFO) lets you do flanging, chorusing, true vibrato, slap-back echo and double-tracking—all from one module! The fantastic versatility comes directly from the SAD-4096 used in it. This chip sports an extra long 4096 "buckets" and can be clocked to give delays from 1 msec on up to 250 msec. Front panel jacks and controls include:

- Input jack (low level to line level)
- Level control
- Clipping LED indicator
- Bright switch (gives a treble "sheen" to the sound)
- Polarity switch (positive or negative flanging)
- Blend control
- Regen control
- ±5V CV input (good for synth work)
- 0 to +10V CV input
- Range control
- Center control
- Volume Control
- Output jack

Note that this is a general purpose voltage controlled module and can be used with just about any control source you can imagine. Since you are not "locked in" to an on-board LFO (which is typically a triangle or sine wave generator—not good for time delays!), you can experiment with a wide variety of modulation techniques, from foot pedals to synth LFOs to "hypertriangular" VCO patches.

Best of all, the **Voltage Controlled Analog Delay** features compansion for low noise operation, uses the classic low noise 4739 dual op-amps, and incorporates all sorts of subtle and advanced techniques to make the signal processing as transparent and quiet as possible.

SUCCESSFULLY BUILDING IT

The **Voltage Controlled Analog Delay** is an extremely sophisticated and complex circuit. Nonetheless, it features a robust design which can be easily built, as long as you employ neat shop techniques and are patient. The name of the game here is:

- Read over this *Information Packet* completely before beginning to construct the circuit. Also, review the schematics until you are thoroughly familiar with them. The better you understand the circuit operation of the **Voltage Controlled Analog Delay**, the more likely you will be able to get it working right off the bat.

- Always discharge any static electricity from your body before handling the chips. The SAD-4096, 4046 and 4041 chips can be ruined by stray static.

- Don't rush. Take your time and check each connection as you make it.

- Be neat. The **Voltage Controlled Analog Delay** contains both audio and radio frequency (RF) portions. Messy wiring will allow these two portions to interact, which will either lead to noisy results or may even cause the circuit to fail altogether.

- Obey all grounding and shielding practices described later in this *Information Packet*. Incorrect or sloppy grounding may lead to hum and other problems.

- Mind your soldering technique. Use rosin core solder manufactured for standard electronics applications and check the finished board for any bridges or blobs which could seriously damage the **Voltage Controlled Analog Delay** when it is first powered up.

PARTS INVENTORY AND IDENTIFICATION GUIDE

Check that your kit has the following items:

- complete chip set, containing:
  - 3 4739 dual low-noise op-amps
  - 1 1458 dual op-amp
  - 1 NE570 compander
  - 1 SAD-4096 bucket brigade chip
  - 1 4046 CMOS phase-locked loop
  - 1 4041 quad TTL buffer
- 1:1 printed circuit board artwork
- rack panel decal artwork
- schematic packet (3 sheets)
- parts placement guide
- this *Information Packet*, containing:
  - circuit description
  - parts list
  - suggested 1U rack panel layout
  - rack panel drilling guide
  - circuit tracing guide

CIRCUIT DESCRIPTION

The schematic for the **Voltage Controlled Analog Delay** is spread across three sheets. These have been provided as a separate packet. Refer to Sheet 1 now.

An electronic musical instrument is patched into jack J1, and then buffered by op-amp IC1a set up as a voltage follower. This is one-half of the dual low-noise op-amp, the 4739. The chips appearing a little later in this circuit operate best with signal levels of around 2V p-p, so potentiometer R44 lets you set the operating level. A Peak LED indicator, to be described shortly, monitors the input signal. If it begins to flash too often, simply dial R44 down a bit. Thus, by means of the Level control and the Peak LED indicator, the **Voltage Controlled Analog Delay** can accommodate most any instrument. By the way, the input impedance of the circuit is a nominal 500K—plenty high enough to prevent instrument loading which can attenuate your high end response.

The output of buffer IC1a splits into three separate paths now. First, the signal is sent directly to one side of the Blend control, R43. This provides the "dry" side which is mixed in with the processed side to generate flanging and chorusing effects. The buffer also feeds the Bright switch and its associated circuitry. We'll touch on these two paths later. But for now, let's trace the input as it moves on to the heart of the module.

The output of IC1a, at pin 13, is AC coupled by means of C30 into one-half of IC5, the NE570 compander chip. This half has been configured as a compressor. Its duty is to compress the dynamic range of your instrument. This gives more predictable results as the signal passes through the bucket brigade chip, and when coupled with the associated expander (coming soon!), helps push the noise level down.

A detailed description of how the compressor works is beyond the scope of this *Information Packet* (see the NE570 data sheet for full details). But notice how the chip's internal VCA is in the feedback loop of the internal op-amp. This allows the gain of the device to be varied under control of the internal precision rectifier. In a nutshell, when the input level starts to rise, the rectifier responds by allowing more signal to pass through the VCA. Since this is in a negative feedback loop, the gain is pulled down. The net effect is that low level signals are boosted, while high level signals are attenuated. What comes out of the NE570 is a waveform whose dynamic range (peaks and valleys) has been reduced by a factor of 2:1.

By their very nature, bucket brigade chips suffer from a disorder known as "aliasing" or "fold-over distortion." This is caused by high frequency components in the input signal interacting with the clocked components of the bucket brigade chip. The way to get around this noxious form of distortion is by limiting the high end of the signal feeding the chip. So, C6 (also in the feedback loop of the NE570's internal op-amp), acts as a simple lowpass filter.

The network of R29, R30 and C38 looks a little weird, but don't let it bother you too much. This is merely a stabilized DC path for the feedback loop of the NE570's internal op-amp. The data sheet demands we include it—so we do, and that's the end of that!

C28 is the filter capacitor for the compressor's precision rectifier. Its purpose is to smooth out the ripple of the rectified signal. Now about that huge resistor, R47. As mentioned above, a compressor tries to "turn down" a hot signal and conversely "turn up" a low level signal. If the signal is really tiny, then the compressor attempts to crank up the gain tremendously to compensate. This can lead to extra noise, of course, but also weird transients. So R47 deliberately feeds in a bit of a positive offset just to keep the compressor from going into overdrive on low level signals. The notion here is that any time a signal is really that small, then it's probably supposed to remain that way.

Now about that Bright switch mentioned earlier; this is implemented in a really slick way. Imagine that S1 is closed. Then you can see that the op-amp internal to the NE570 is really acting as a mixer; two lines are combined at its inverting input. One passes through C30 and the 20K resistor inside IC5a, while the other passes through C12 and R4. Now observe that C30 is quite large (2.2µF), and so offers very little impedance to any audio frequency. But C12 is quite small (0.0047µF) and tends to block low frequency components. Moreover, R4 is 20 times smaller than the internal 20K resistor, meaning that this path is given 20 times "more weight." The net effect? When S1 is open, all audio frequencies are passed equally. But when S1 is closed, an extra boost is given to the high end. The sound thus generated has more of a sparkle or sheen to it, hence the name Bright for this switch.

Incidentally, there is one more path to the inverting summing node of the op-amp inside IC5a, which comes from the Regen control. We'll stall off a description of this until we get a little further along in the circuit.

Due to the way the NE570 is implemented and also because it runs off of a single +15V power supply, the output at pin 10 biases up to about +6VDC. Unfortunately, the input of IC6, the SAD-4096, needs to be biased at about +3V. So, we have to go through one extra op-amp stage to fix things. First, notice that the output of the compressor is AC coupled to IC1B by means of C32. This blocks the +6V bias mentioned earlier.

Since R35 and R34 have a ratio of unity, the amplitude of the audio signal passing through IC1b is left unchanged. But trimmer R13 injects a negative offset through R41. Since the op-amp is configured as an inverting amplifier, this will be transformed into a positive bias at the output. Thanks to R41 (which sets the basic gain of the offset) and to R5 (which limits the low end of the trimmer), the DC bias at the output of IC1b will be about +3V when trimmer R13 is near its mid-rotation. Adjusting this lets you optimize the bias to the SAD-4096 which minimizes the distortion.

In case you're wondering, we have gone through all of this rigmarole with IC1b just to make sure that we're driving the SAD-4096 directly from the low impedance output of an op-amp. This gives far superior results than the usual passive component biasing scheme used in cheaper circuits.

And finally—we're ready to send the signal to that magical bucket brigade chip, the SAD-4096. The output of IC1b connects to pin 14 of IC6 by means of R2. We've been assuming that the signal up to this point is at a max of 2V p-p, and then of course IC1b DC biases it up to +3V as well. But as added protection, D1 will kick into action if the input at pin 14 of the SAD-4096 drops below zero for any reason. It'll dump the potentially harmful negative voltages to ground. R2 is a series limiting resistor to keep D1 from frying in this case.

The properly biased audio signal is now inside the SAD-4096, bouncing down the buckets, one after another. A clock circuit controls how long it takes it to do this, and will be described later. Once the signal passes down the line, it exits at pins 8 and 9. There are two outputs for a very important reason. Thanks to the way the SAD-4096 has been designed, any residual clocking signal will appear on these two outputs in a differential form (one positive going, one negative going). R7 sums these in a simple fashion, and by properly adjusting this trimmer, it is possible to null out or minimize the unwanted clocking pulses.

If you take the time to study the SAD-4096 data sheets, then you'll find out that what ends up at the wiper of trimmer R7 is in fact a "stair-step" approximation to the original waveform. Even though the steps are quite tiny, we need to even them out and get a smoother approximation to the desired analog input. C13 to ground from the wiper of R7 is a step in the right direction. This is essentially a simple low-pass filter; the sharp edges of the stair-step approximation are smoothed down a bit.

But we have to get more serious than that if we want to obtain good noise and distortion figures. So, IC2a and associated components pull together to make a fairly decent 2nd order low pass filter. Called a Sallen-Key unity gain filter, it does a fantastic job of eliminating any high-end garbage as well as smoothing out the stair-step approximation of the waveform. With the values of R20, R21, C7 and C11 shown, it has a -3dB cutoff frequency of about 7KHz. This is just low enough to permit good audio bandwidth, and yet do the job it's supposed to. If you'd like to mess with the filter, then by all means see Don Lancaster's *The Active Filter Cookbook*, (Howard W. Sams: Indianapolis, 1978), which presents all the design equations in an easy to use form.

Okay, so we've compressed the audio input with IC5a, delayed it with IC6, and are now ready to expand it once again. By expanding the dynamic range, we not only restore the signal to its natural form, but also push the noise floor down by about 45dB. When the signal leaves the filter at pin 1 of IC2a, it then moves on to the input of the expander implemented by means of the second half of IC5. This takes us to Sheet 2 of the schematic now.

The processed signal is injected into IC5b at pins 2 and 3. You'll note now that the internal rectifier directly controls the internal VCA. Thus, the louder the input signal, the more gain it is given (making it louder yet). This is exactly the opposite of what the compressor section did.

C29 is the rectifier capacitor, and smoothes out the DC control voltage to the internal VCA somewhat. To keep the VCA from zooming up into ever higher and unpredictable gains, R48 bleeds off a tiny bit of excess voltage from C29. (Are you starting to see a pattern? Again, this is exactly the opposite of what was going on in the compressor section).

Pin 6 of IC5b connects directly to pin 7. Observe that this has the effect of putting the internal 20K resistor into the feedback loop of the internal op-amp, which sets the basic gain. C8 is in parallel with this loop; its job is to lowpass filter the signal just a bit more.

This is probably a good time to mention that pins 8 and 9 of the NE570 were originally made available to permit trimming the total harmonic distortion of the chip. This would be overkill for the **Voltage Controlled Analog Delay** however. According to the data sheet, these pins need to be bypassed with caps if they're not used, and that's the purpose of C9 and C10.

A really nice feature of the **Voltage Controlled Analog Delay** circuit is the inclusion of a Peak LED indicator. When this flashes, you know that your input signal is too hot and that you need to tame it a bit with Level control R44. (A signal that is too hot may lead to distortion or, worse, could even damage IC5 and IC6). Let's see how it works. R49 taps off of the expander rectifier cap (which will be a DC voltage proportional to the strength of the signal). This then feeds IC4a, which has been configured as a comparator. The voltage on pin 2 of this chip, which is determined by the wiper of trimmer R14, determines the threshold at which the comparator flips "on."

Here's a subtle point to consider. First, notice that the output of comparator IC4a drives LED D3 by way of R8. More importantly, the cathode of D3 connects to -15V, not ground as is so often seen in many electronic music circuits. The reason for this is simple. Most LEDs lead happier lives if they are not reverse biased by large voltages. With the arrangement just described, there's either a +30V differential from pin 1 of IC4a to the cathode, or else there's a 0V differential (and never a negative differential). Of courses, R8 limits the current flow to a safe level.

You might wonder why the lowly 1458 op-amp is used here. After all, isn't this a cheesy chip unsuited for low noise electronic music applications? Well, not here! This half of the chip is only doing control voltage things (as is the second half, yet to come in the description), so why not use it and save money!

But back to the expander. The output is available at pin 7. C34 blocks the DC bias, while permitting the audio (AC) to ride on to the polarity changer configured around IC2b and associated components. The purpose of this section is to let you either add or subtract the time delayed signal from the dry signal. What good is that, you might ask? As it turns out, these two variations end up sounding very different from each other and both are musically useful. For example, consider flanging with the **Voltage Controlled Analog Delay**. Positive flanging gives a warm, smooth and mellow sound, while negative flanging leads to a sharper, metallic effect. This versatility is something we've got to have! And best of all, it only takes a toggle switch, an op-amp and a handful of passive components to implement. Let's see how.

As mentioned, the expander output is coupled to the polarity changer by means of C34, which then goes to the wiper of switch S2. If S2 is in the negative position, then op-amp IC2b is configured as an inverting amplifier. In this case, R23 is unused (no connection to it on the input side), while R24 pulls the noninverting input to ground. The ratio of R22 and R31 sets the gain of the circuit at 1/2.

Now consider the circumstances when S2 is flipped to the positive side. R31 is not used and can be ignored. R22 directly couples the output back to the inverting input, making this a voltage follower configuration with a gain of 1. However, the input signal is applied to the voltage divider composed of R23 and R24. These are equal valued resistors, so the level is chopped in two, and then buffered by IC2b. Again, the net gain is 1/2.

Incidentally, when operating as a buffer, the high impedance of IC2b's non-inverting input can act as a magnet for RF interference. So, we plop in C4 to dump any rubbish to ground. Similarly, C3 guards the circuit when it has put on its hat as inverting amplifier.

So the audio signal is either inverted or left alone, depending on the position of S2, and in both cases it is attenuated by half. The output at pin 13 of IC2b then splits, going to the Regen control and the Blend circuitry. Let's take on the Blend section first.

C36 AC couples the signal to inverting amp IC3a. Notice that the ratio of R32 and R25 is about 2, which sets the gain at 2 as well. The reason for this should be obvious now; the polarity changer halved the signal; now the Blend driver doubles it again. The net effect is a strict unity gain. By the way, C2 which parallels feedback resistor R32, helps dump any radio frequencies which might have been hitching a ride. The circuit acts as a simple low pass filter with a cutoff frequency of about 33KHz.

The output of IC3a then feeds one side of R43, the Blend control. The other side of this pot comes from input buffer mentioned earlier. The wiper of R43 then can pick off any mix of these two signals (dry or processed). Further notice that both sides of R43 are being driven directly by op-amps, which is what leads to it having such a smooth feel—you can rotate the pot from full dry signal continuously to equal mix on up to full delayed signal.

The wiper of R43 presents a rather high output impedance, so IC3b buffers it. This is standard non-inverting amplifier, with the input at pin 9. The gain is about 5, just to give the **Voltage Controlled Analog Delay** a bit of a boost (1+R26/R9). Once again a capacitor, C5, comes to the rescue to help dispose of any unwanted RF or high frequency noise. The output is AC coupled by C37, and then applied to Volume control R16. Dial this down as desired to supply a final output to jack J2.

As mentioned earlier, the output of the Polarity changer also goes to the Regen pot, R15. By cranking this up you can get multiple echoes or an intensified chorus/delay effect depending on the delay range you've selected. The wiper of R15 (the Regen control) is AC coupled by C27 right back into the compressor by means of R17. Setting the value of C27 down around 0.22µF impedes the bass frequencies somewhat. This helps keep the unit from "booming" in an unpleasant manner.

At this point we've covered the audio section of the **Voltage Controlled Analog Delay** in quite some detail. It's now time to consider the controller aspects, so turn to Sheet 3 of the schematic.

Many users will want to modulate their **Voltage Controlled Analog Delay** with an LFO waveform from an analog synthesizer. The usual output for LFOs is a triangle wave which moves from -5V on up to +5V and back again. This type of control voltage can be applied directly to jack J3. IC4b is configured as a unity gain inverting amplifier. Notice, however, that R42 sums in a constant offset of about -5V. (That's given by -15V times R38/R42). Since this will be inverted by the op-amp, what comes out at pin 7 will be a triangle wave that is shifted upward by +5V. To put it another way, the triangle now sweeps from 0V on up to +10V and back again.

The manipulated waveform is "normalled" to jack J4. But by inserting a plug into J4, you can override this default. Use J4 when you wish to modulate the **Voltage Controlled Analog Delay** with a 0V to +10V control signal (for example with foot pedals, some brands of synths, and especially homebrew modulators). Notice also that the responses of J3 and J4 are inverted with respect to each other. This gives you some additional versatility—simply use the input that sweeps the delayed effect in the "right direction" for your application.

Range control R39 lets you tame the amount of modulation coming in from either of the control inputs. D2 is wired in reverse, right on the back of the pot. Its duty is to keep any harmful negative voltages from being coupled to later circuitry. A negative voltage will forward bias the diode, dumping the charge directly to ground. R6 limits the current flow to a safe value.

By the way, if you decide to not plug anything into either J3 or J4 (for straight echo effects), then turn down the Range control completely. This will keep that +5V offset in IC4b from messing up the "feel" of the Center control (to be discussed next).

The wiper of the Range control heads to the CV input of the high frequency VCO built around IC7. This is the 4046 CMOS phase-locked loop chip. Also feeding the control voltage input at pin 9, is an offset from Center control R40. You spin this knob to knock the delay time up or down some fixed amount. In other words, the Range control determines the "width" of the total sweeping action, while the Center control determines the central point of the sweep.

Trimmer R33 lets you adjust the longest delay available from the unit. As a general rule, the longer the delay the more noise you'll have to put up with. So, set this trimmer to strike the compromise you like best between noise and long echo effects.

C1 and R12 establish the basic frequency range of IC7. With the values shown, the VCO can be swept from around 10KHz on up to about 1.5MHz (depending on the control voltage at input pin 9).

The SAD-4096 needs to be driven by complementary clock pulses. Furthermore, these pulses need to have quite a bit of "oomph" to them. That's because the two clock inputs to the SAD-4096 each has an input impedance of 1000pF. That's an awful lot of capacitance to overcome, so we direct the output of IC7 to a CMOS/TTL buffer, IC8. As you can see from the block diagram of this chip, it contains four inverter/buffers. The inverters give us the complementary clock pulses we need, and by ganging the proper outputs of the four pairs, we also obtain a large amount of drive. That's perfect for overcoming the hefty input capacitance of the

SAD-4096! The complementary pulses (ϕ1 and ϕ2) then head straight to the clock inputs of the SAD-4096.

To wrap things up, notice the array of power supply capacitors shown on Sheet 3. C39 and C40, which are 10μF electrolytic capacitors, decouple the entire circuit from your power supply. Capacitors C14 through C21, on the other hand, attach very close to the respective power pins of the op-amp chips. It's their duty to bypass the power lines coming in to each IC. This keeps RF and other garbage from hitching a ride either to or from the op-amps. It need hardly be said that in a complex circuit of this nature (containing both audio and RF signals), it is mandatory to include these bypass caps to get the best performance possible.

And that concludes your rather lengthy tour of the inner workings of the **Voltage Controlled Analog Delay** line! If it all makes sense to you, then you're ready to start building!

HOW TO BUILD IT

Owing to the complexity of this circuit, (and the fact that it contains both audio and RF portions) don't even consider building it in any other way than with a printed circuit board. Included with this kit is 1:1 printed circuit board artwork. You can use this along with any of those neat iron-on sheets to transfer the pattern to a copper clad board. (Ads for these transfer sheets appear frequently in the back of most electronics magazines). By the way, we've duplicated the artwork twice on the enclosed sheet—this will give you a second chance in case you mess up on the first go!

Your next step is to gather up the parts. You already have the hard-to-find ones (the chips). All of the others can be obtained from just about any electronics mail order house. The complete parts list is shown on page 8.

When gathering up the parts, don't be tempted to substitute capacitor types. All of the critical frequency tuning ones must be polystyrene or mylar for maximum stability. The others are either electrolytics or disc types, since they aren't involved in any precise tuning operations.

Also observe the taper of the potentiometers. These have been carefully chosen to give the controls the "right feel." Substituting here can make the unit hard to use.

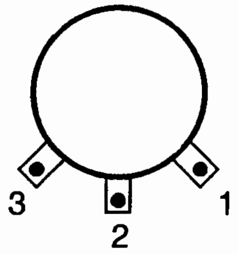
The Parts Placement Guide provided elsewhere shows you how to load the board. Pay close attention to the polarities of the diode and the electrolytic capacitors; one false move there can spell disaster! And of course, make certain you've oriented the chips in their sockets properly.

The Voltage Controlled Analog Delay can be built behind a 1U rack panel. If you decide to go this route, then you might find the suggested layout and drilling guide on page 7 helpful. (Blank 1U rack panel kits are available from *Midwest Analog Products*. See the Parts List on page 8 for details). A separate sheet included with this kit has the artwork for some decals you can make. Simply take the sheet to your local photocopy house and have them copy it onto some clear sticky-back material. Then cut the decals out with an X-acto knife, peel off the backing, and stick them to your painted panel. Apply several very light layers (not too thick or the ink might run) of clear plastic spray paint to protect the decals.

To complete wiring the board to the front panel, use the Parts Placement Guide provided elsewhere. A number of connections to the panel must be made with shielded wire. This is essential for low noise performance. But notice the following carefully: in all cases the shield or braid is grounded at the circuit board, but is left to float on the panel end of the wire. This is vital to avoid ground loops which can cause hum. If you examine the circuit board, you will note that each connection which requires shielded wire always has a ground pad next to it. So, for emphasis, let's say this again: connect the hot wire of a shielded cable to the required pad, then connect the shield to the ground pad right next to it. Connect the other end of the hot wire to the proper front panel control lug, but do not connect the shield at this end to anything.

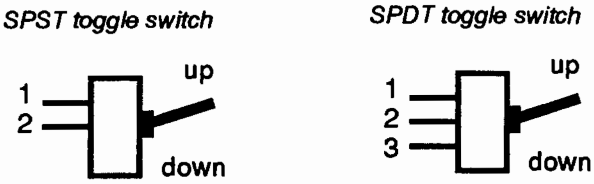
The anode and cathode pads for LED D3 are marked on the parts placement guide with the letters "a" and "c", respectively. To keep things tidy, you might want to run these to the LED with a twisted pair.

The lugs of the potentiometers are called out on the Parts Placement Guide by the numbers 1, 2 or 3, according to the illustration below. For example, the pad labeled "R15/2" tells you that a wire should connect from it to lug number 2 of R15 (the Regen control).



The three solder lugs of the potentiometers as seen from the back. The shaft is on the other side, and isn't visible here.

For convenience, the lugs on the two switches are also numbered. The following illustration shows the arrangement.



When the toggle of the SPST switch is up, the switch is off. When it is down, terminals 1 and 2 are connected. When the toggle of the SPDT switch is up, terminals 2 and 3 are connected. When it is down, terminals 1 and 2 are connected.

Finally, you will need to apply a source of bipolar 15V to power up the **Voltage Controlled Analog Delay** module. (See page 10 for details on a high quality power supply you can build.) You will find the +15V, -15V and G (ground) pads grouped close to each other on the printed circuit board. Observe that there is one additional GND pad near these. This pad is used to supply a ground run to the front panel, where it daisy chains as required to the various pots and jacks. And here's an important point: this is the only place where ground connects to the front panel. To avoid ground loops (a leading cause of hum), it is vital that the front panel controls connect to ground in one place only.

If you should need to do any troubleshooting or debugging (or perhaps just confirming you've made a proper hookup), then the Circuit Tracing Guide on page 9 should prove useful. Notice that this shows the basic positions of the parts with respect to the copper foil traces on the printed circuit board. However, the polarities of the electrolytic caps aren't shown here; refer back to the Parts Placement Guide to check these.

TWEAKING THE TRIMMERS

When you first fire up the **Voltage Controlled Analog Delay**, you'll probably hear nothing or else a bunch of distorted garbage. Don't freak out; this module simply won't perform right until the trimmers have been properly set. Fortunately, this is an easy process. Start by patching an instrument into J1, and your amplifier from J2.

> **Caution!**
> Until the trimmers have been properly set, some extremely loud and unpleasant sounds are possible. So, protect your ears and amplifier by keeping careful tabs on the volume control.

To proceed with the calibration, turn all trim pots to their mid positions, then set the front panel controls as follows:

Level:	mid
Bright:	on
Regen:	full off
Polarity:	+
Blend:	full delayed
Volume:	as required
Range:	full off
Center:	full off

Now power up everything and play your instrument. Adjust the bias trimmer (R13) until you can hear a delayed replica of your instrument. In some positions of this trimmer, you'll hear nothing and in others you'll hear a severely distorted version. Tweak the trimmer to minimize the apparent distortion. Now start easing up the Level control and readjust R13 as required. Continue doing this until you are able to pass the hottest signal possible without distortion.

Then set the Peak LED trimmer (R14) until the LED starts to flash on the loudest passages you play. In other words, the LED will light on levels that are just starting to move you into the distorted region.

Now adjust the max delay trim until you can hear the high frequency clock signal bleeding into the audio. Make this as high pitched as you can discern. (You deliberately want to hear the clock signal for this step; you'll reset the max delay trim in just a minute). Now adjust the clock null trim (R7) to minimize or eliminate the clock signal feeding through. If you wish, you can also carry out this step efficiently on an oscilloscope; simply trim out the clock edges from the audio signal.

Finally, set the max delay trim (R33) to an acceptable value for your purposes. Remember, the longer the delay the more the noise, so don't set this any higher than you actually intend to use. Be forewarned: the **Voltage Controlled Analog Delay** is capable of some extremely long echoes, but the hiss and hash coming from the SAD-4096 will drive you nuts! For cleanest results, consider this more of a chorus/flanger unit and stay away from long echoes.

Now, start playing with the controls to get familiar with them. After you find some pleasing combinations, and have used the unit for a little while, go back and adjust the trimmers again. In general, electronic parts need to "settle in" just a bit. And you'll probably find that you can optimize the trimmers for some of your favorite front panel settings. So, don't be shy about readjusting the trimmers more than once to obtain the optimal settings.

MODS AND TIPS

The **Voltage Controlled Analog Delay** opens up all sorts of neat possibilities for audio processing. Thanks to its "open-ended" control inputs which don't lock you into any one control source, lots of possibilities may occur to you. The WWW page of Midwest Analog Products will serve as a clearing house for these ideas. If you come up with some new application, then be sure to send it in. We'll post it for other users to benefit from. And we'll be listing a few ideas of our own, too, so be sure to tune in frequently and see what's new!

Here are a few brainstorms to get you started:

1. If you are going to experiment with long time delays for echo effects, then you'll want to drop the cutoff frequency of filter IC2a accordingly. It's set for about 7KHz with the components shown. But you can move it down to 3.5KHz (which will cut out quite a bit of noise) by doubling C11 and C7 to 0.0047µF and 1200pF, respectively.

2. R17 keeps the Regen control from bumping into feedback under most conditions. If you'd like to push the Regen a bit more for even deeper effects, then decrease this resistor. Start with 10K and see what happens. But remember, that feedback can be loud, so watch the volume control on your amp!

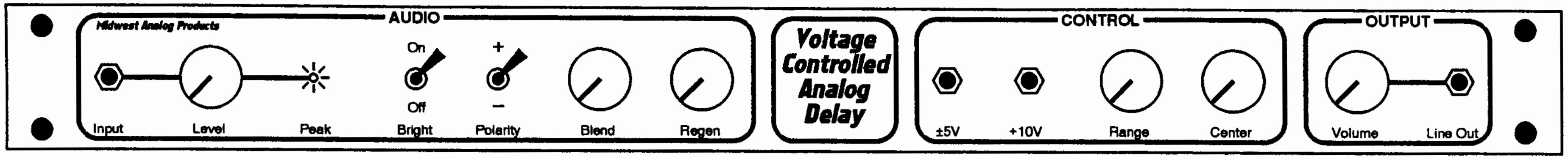
3. R45 and R46 mix the Range and Center controls in a rather simple fashion. You could replace this section of the circuit with a better mixer fashioned around a simple op-amp stage.

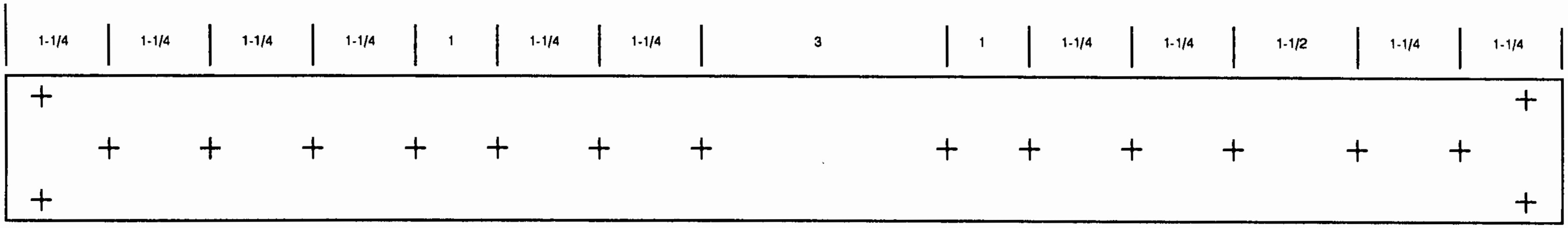
4. Triangle and sine waves really don't modulate analog delays in the best way possible. That's due to the inverse relationship between the clock frequency and the time delay period. The bass range tends to get passed over too quickly, and things drag on way too long in the treble range. So, you might want to try experimenting with Craig Anderton's idea of "hypertriangular" modulation. This is possible with most any exponential synth VCO. See his two part article, "Building the Hyperflange+Chorus," *Modern Recording & Music*, July 1983 and September 1983, for details.

ACKNOWLEDGMENTS

As near as I can determine, only one other person has ever designed a delay circuit using the SAD-4096, specifically with the DIY electronic musician in mind. You won't be surprised to learn that it came from the prolific Craig Anderton! He described his results in the article, "Build a Chorus-Delay," which appeared in the January 1982 issue of *Guitar Player*. Several of his ideas in that article influenced my design, and I'm grateful to acknowledge it here. I also want to thank Craig for his friendship and encouragement over the past two decades.

The idea of using the 4041 quad TTL buffer to drive the stiff clock inputs of a delay chip is apparently due to Dave Tarnowski. I've never met Dave, but remember eagerly reading some of his results in *DEVICE Newsletter* back in 1971. (Ah...how I miss those days when we DIY-ers had several magazines all our own—now we have none.)





Suggested 1U Rack Panel Layout and Drilling Guide
All dimensions in inches.

PARTS LIST

All resistors are 1/4-watt, 5% values.

R1 - R3	10Ω
R4 - R6	1K
R7	1K trimmer potentiometer
R8	3.3K
R9	4.7K
R10 - R12	10K
R13, R14	10K trimmer potentiometer
R15, R16	10K audio potentiometer
R17	12K
R18 - R26	22K
R27	33K
R28	39K
R29 - R32	47K
R33	50K trimmer potentiometer
R34 - R38	100K
R39, R40	100K audio potentiometer
R41	220K
R42	300K
R43	500K linear potentiometer
R44	500K audio potentiometer
R45, R46	1M
R47 - R49	10M

Semiconductors

D1, D2	1N914 or 1N4148 diode
D3	red LED
IC1 - IC3	4739 dual low-noise op-amp
IC4	1458 dual op-amp
IC5	NE570 compander
IC6	SAD-4096 bucket brigade
IC7	4046 phase locked loop
IC8	4041 quad TTL buffer

All capacitors are 16V or better.

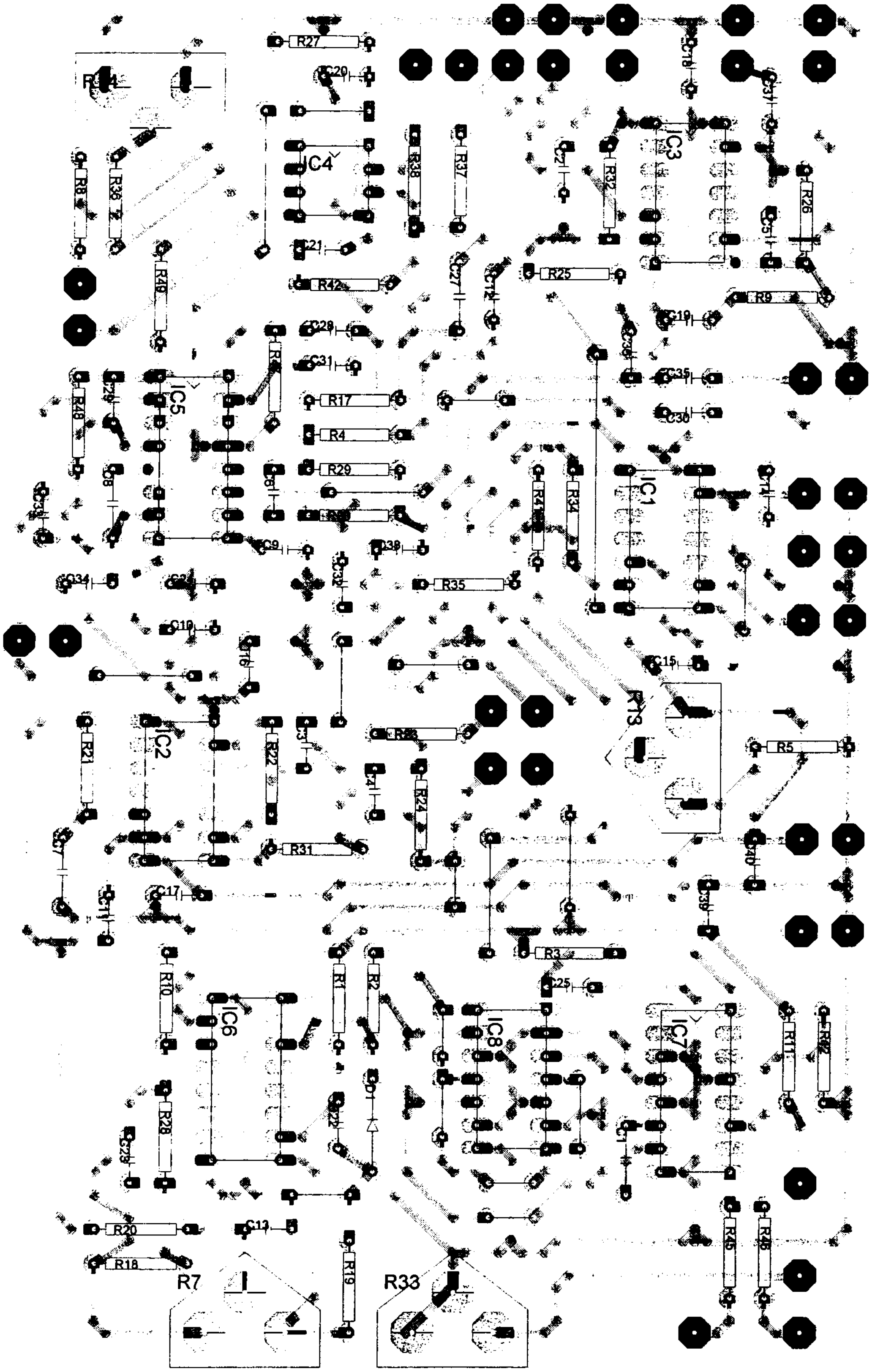
C1	47pF or 50pF polystyrene
C2	100pF disc
C3 - C5	220pF disc
C6	270pF polystyrene
C7, C8	560pF polystyrene
C9, C10	0.001μF disc
C11	0.0022μF mylar
C12, C13	0.0047μF mylar
C14 - C21	0.01μF disc
C22 - C25	0.1μF disc
C26	0.1μF mylar
C27	0.22μF mylar
C28, C29	1μF electrolytic
C30 - C37	2.2μF electrolytic
C38 - C40	10μF electrolytic

Other components

S1	SPST mini-toggle switch
S2	SPDT mini-toggle switch
J1 - J3	1/4" phone jack, n.o.
J4	1/4" phone jack, n.c.

Miscellaneous: printed circuit board, LED holder, IC sockets, front panel, knobs, wire, etc.

Note: At the time of this writing, *Midwest Analog Products* has limited quantities of the polystyrene capacitors in stock (50pF, 270pF and 560pF) at $0.25 each. Please write or e-mail first to confirm availability.



Circuit Tracing Guide

If you run into trouble, or simply wish to modify your **Voltage Controlled Analog Delay,** this guide will let you trace circuit operation with respect to the board. Please note that the electrolytic capacitor polarities are not marked here (C28-C40). To find them, refer to the Parts Placement Guide.

RETURN POLICY

We want you to be satisfied with your purchase. If after you have read this *Information Packet* you decide that the **Voltage Controlled Analog Delay** kit is beyond your abilities, you may return it for a full refund (less your shipping expense). The following restrictions apply:

- You must return the unopened bag of parts (part #K140), this *Information Packet* and all other printed materials in new and unharmed condition.
- No returns accepted if the kit of parts (part #K140) has been opened or tampered with.
- No returns accepted beyond 2 weeks (14 days) of your receipt of the kit.
- Return shipping expense is your responsibility and must be prepaid.
- Damage caused by improper packing is your responsibility.

HOW TO REACH US

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CREDITS

The Voltage Controlled Analog Delay was designed by Thomas Henry during the week of the Vernal Equinox, 1998.

ALSO OF INTEREST

Platinum Power Supplies

If you've been thinking of starting to build a modular system, or wish to upgrade your present one to handle more modules, then have we got a deal for you! Here are the new Platinum Power Supply Kits. These have been specifically designed to be hands down the best power supplies ever for electronic music. But besides all of the "platinum" features (which we'll describe in a moment), let's cut right to the question which is probably uppermost in your mind: "How much oomph is there?" The Platinum Power Supply Kits are capable of delivering a whopping 750mA per output. That's enough juice to run over 37 ADV-SNAREs at once, for example!

The Platinum Power Supply Kits come in two models, a +5V version (PPS1) and a bipolar 15V version (PPS2). The PPS1 can handle all of your digital needs (like our ADV-MIDI and MIDIGATOR projects), while the PPS2 is perfect for powering all of your analog equipment, like the **Voltage Controlled Analog Delay**.

If you've checked around the catalogs, then you know how hard it is to find decent power supplies for electronic music at a fair price. Sometimes you see non-regulated ones—boo! Then other times you see regulated ones, but their outputs are so anemic as to be laughable. The Platinum Power Supply Kits give you power, safety and reliability at a price well within anyone's budget.

The PPS1 puts out +5V at 750mA, while the PPS2 puts out ±15V at 750mA per side. The kits share the following features:

- 110VAC or 220VAC operation (overseas customers take note!)
- exceedingly hefty transformers
- fully regulated voltages
- input and output short circuit protected
- output reverse voltage protected
- thermally protected

Included with each are all of the niceties you would expect, like a heavy-duty power switch with attractive faceplate, power on indicator LED, panel mount fuse holder and fuse, 110AC line cord with plug, beautiful circuit board with solder reflow finish, extremely complete and detailed *Assembly Guide*, heatsinks, hardware, shrink tubing, LED clip—the works!

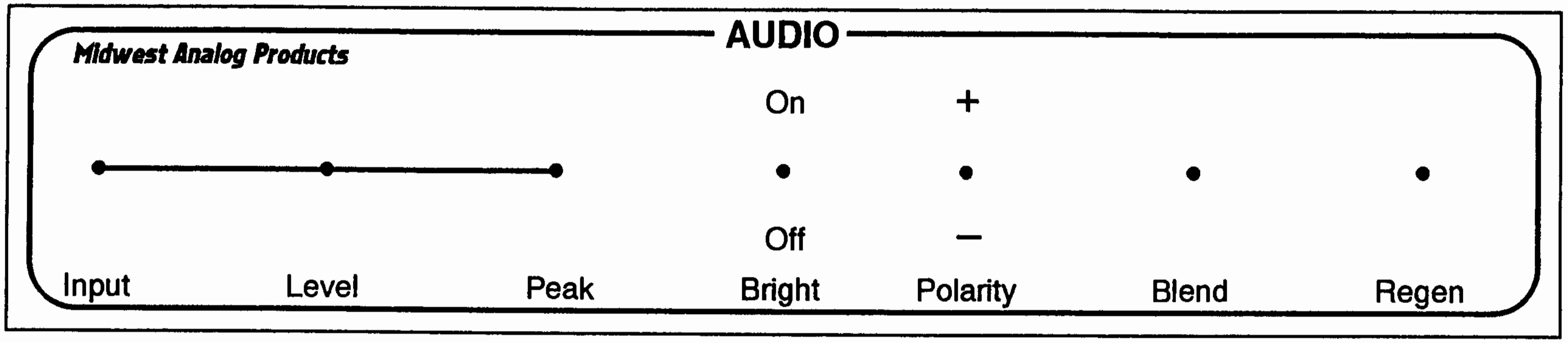
In order to keep our prices down and pass the savings on to you, the Platinum Power Supply Kits do not include rack panels or heatsink grease. Small tubes of heatsink grease are available at Radio Shack for around two bucks. (That's the reason we don't provide it with the PPS Kits—Radio Shack is able to supply it at a much lower price that we would be able to.)

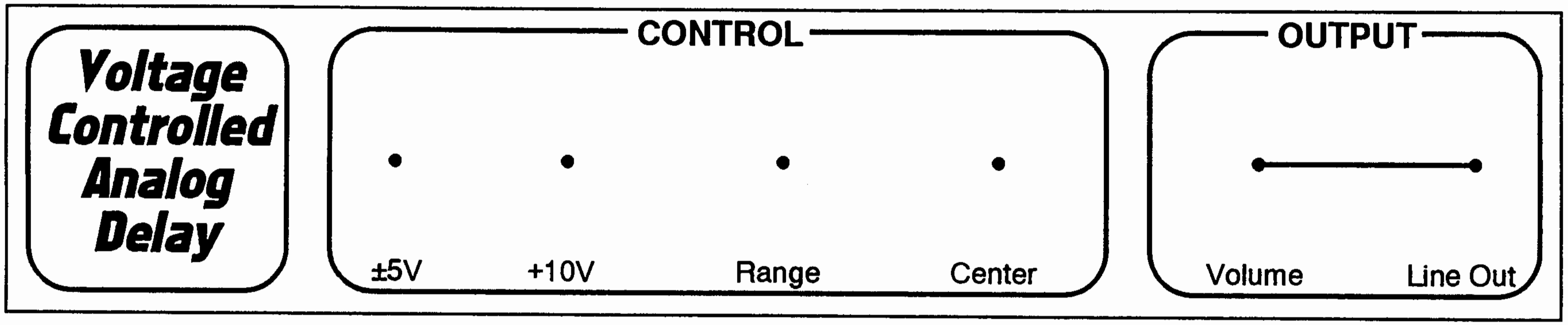
#PPS1 — +5V Power Supply Kit ....Price: $27.95

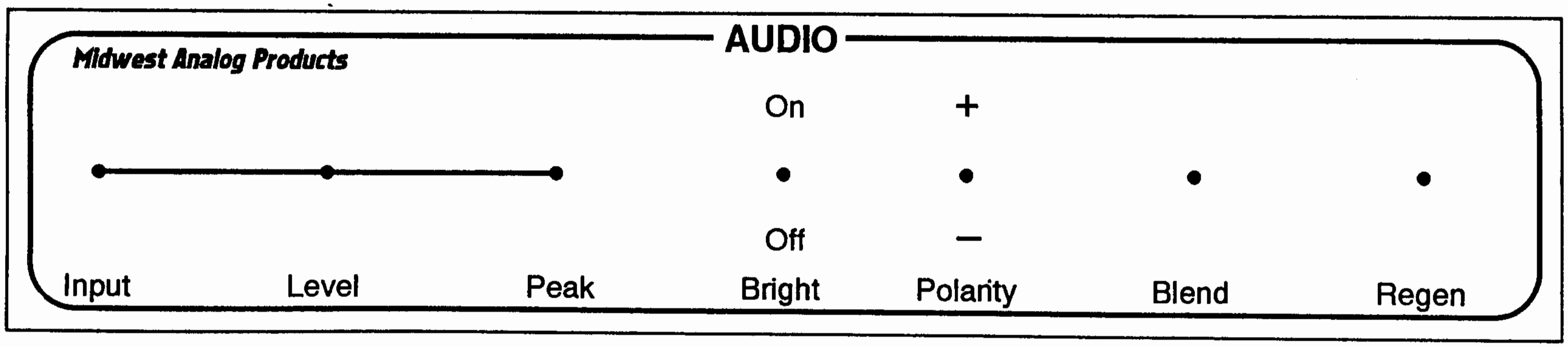
#PPS2 — Bipolar 15V Power Supply Kit ....Price: $44.95

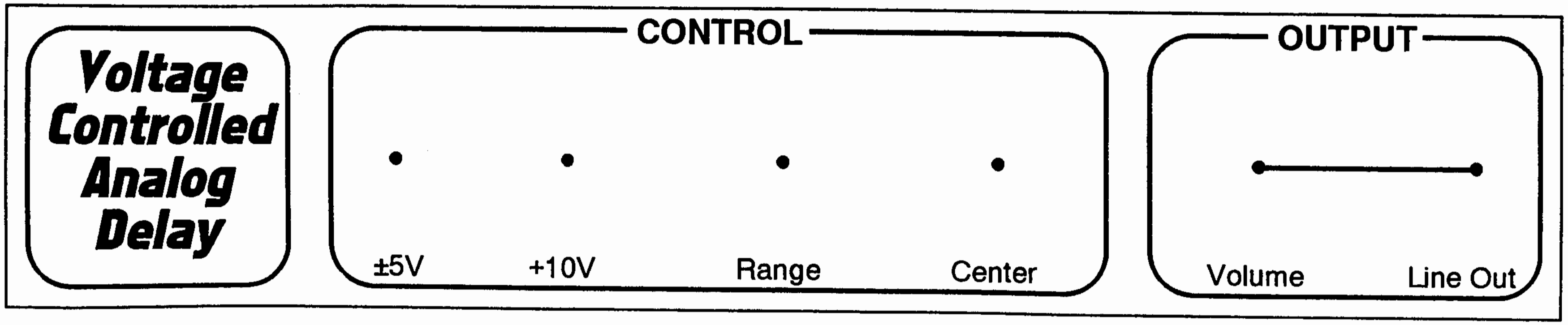
#PPS3 — Combo Power Supply Kit (PPS1+PPS2) ..... Price: $64.95

($15 minimum order. Please add 10% shipping and handling.)









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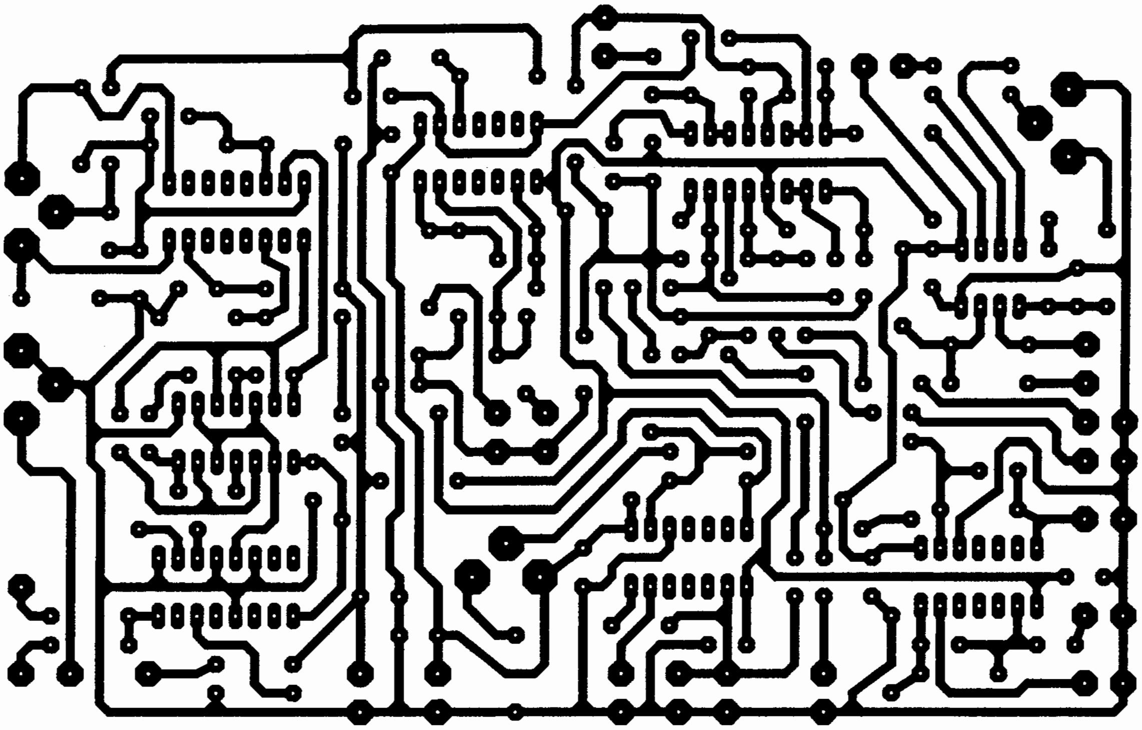

This has been printed as seen "looking through" the board from the component side (which is required by iron-on transfer methods.) The artwork is duplicated here so that you will have two copies—just in case one of them messes up in the process.

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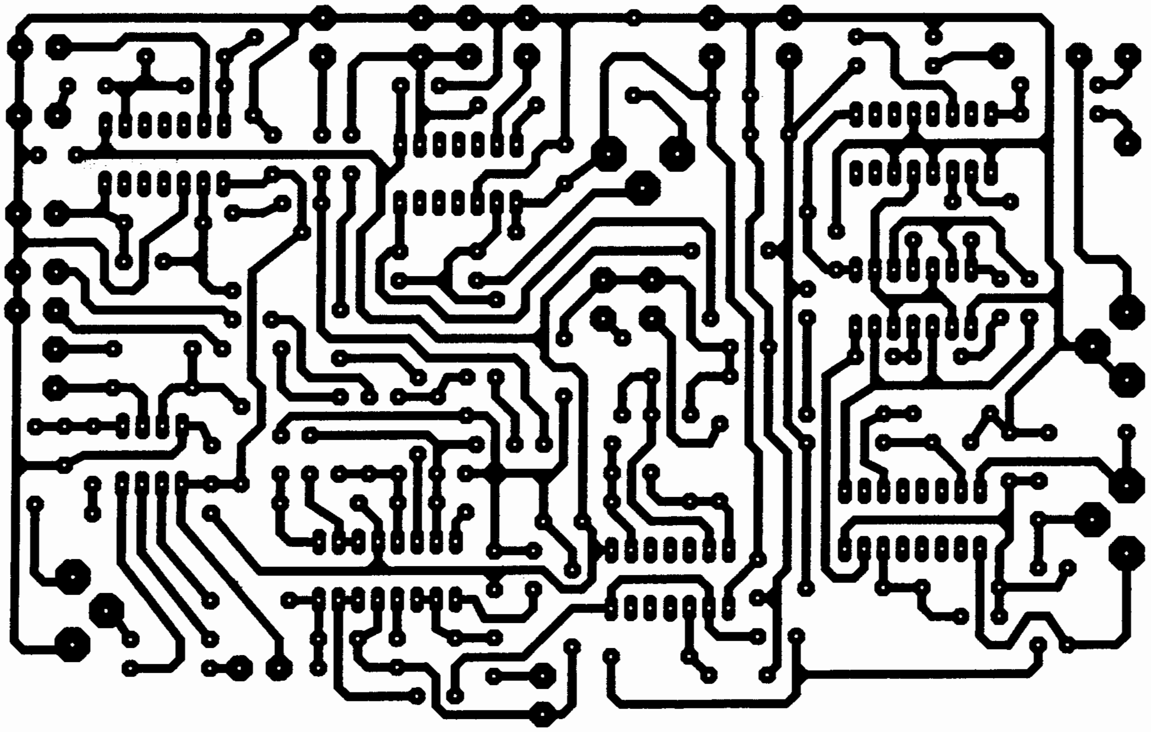

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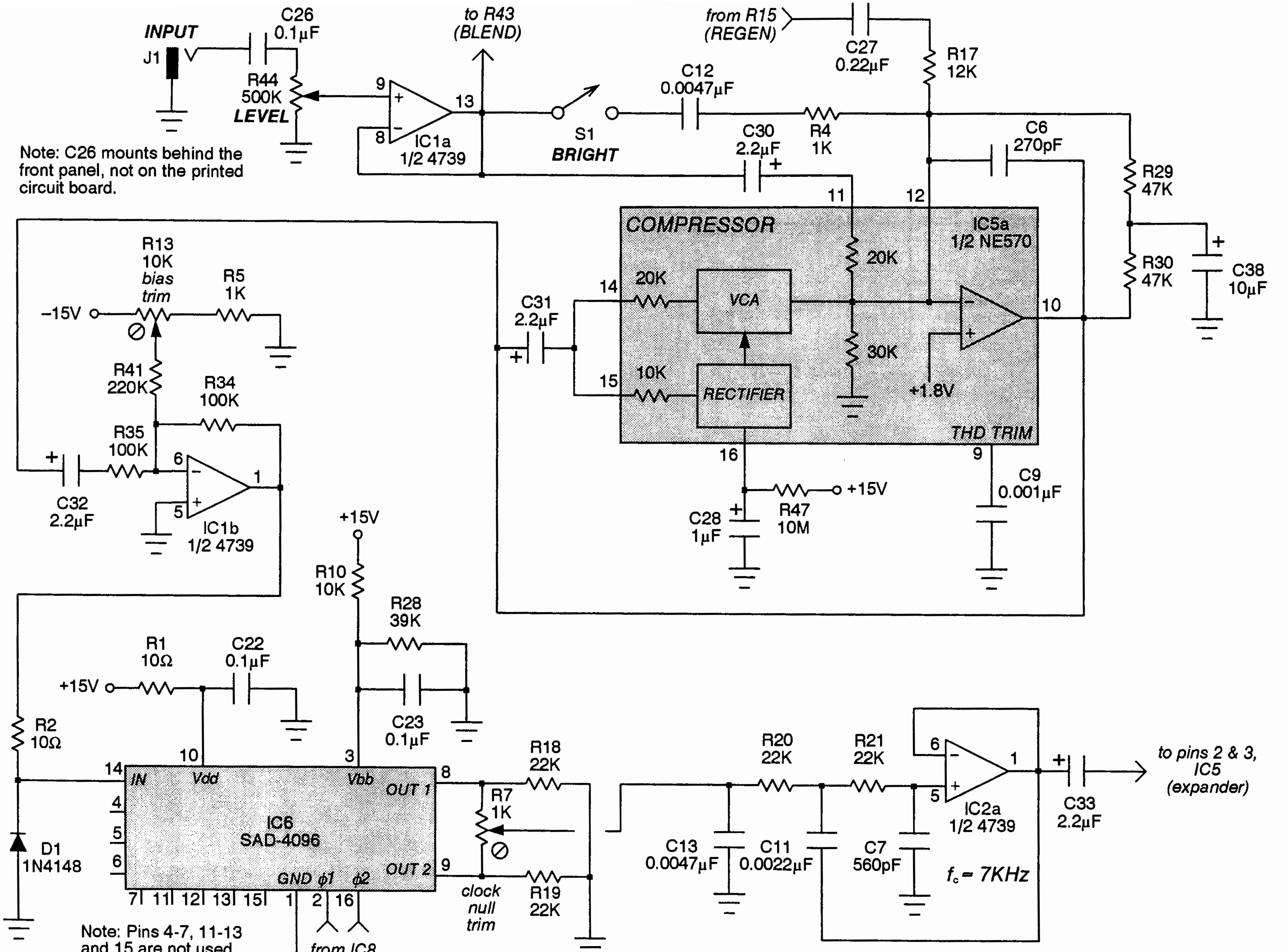
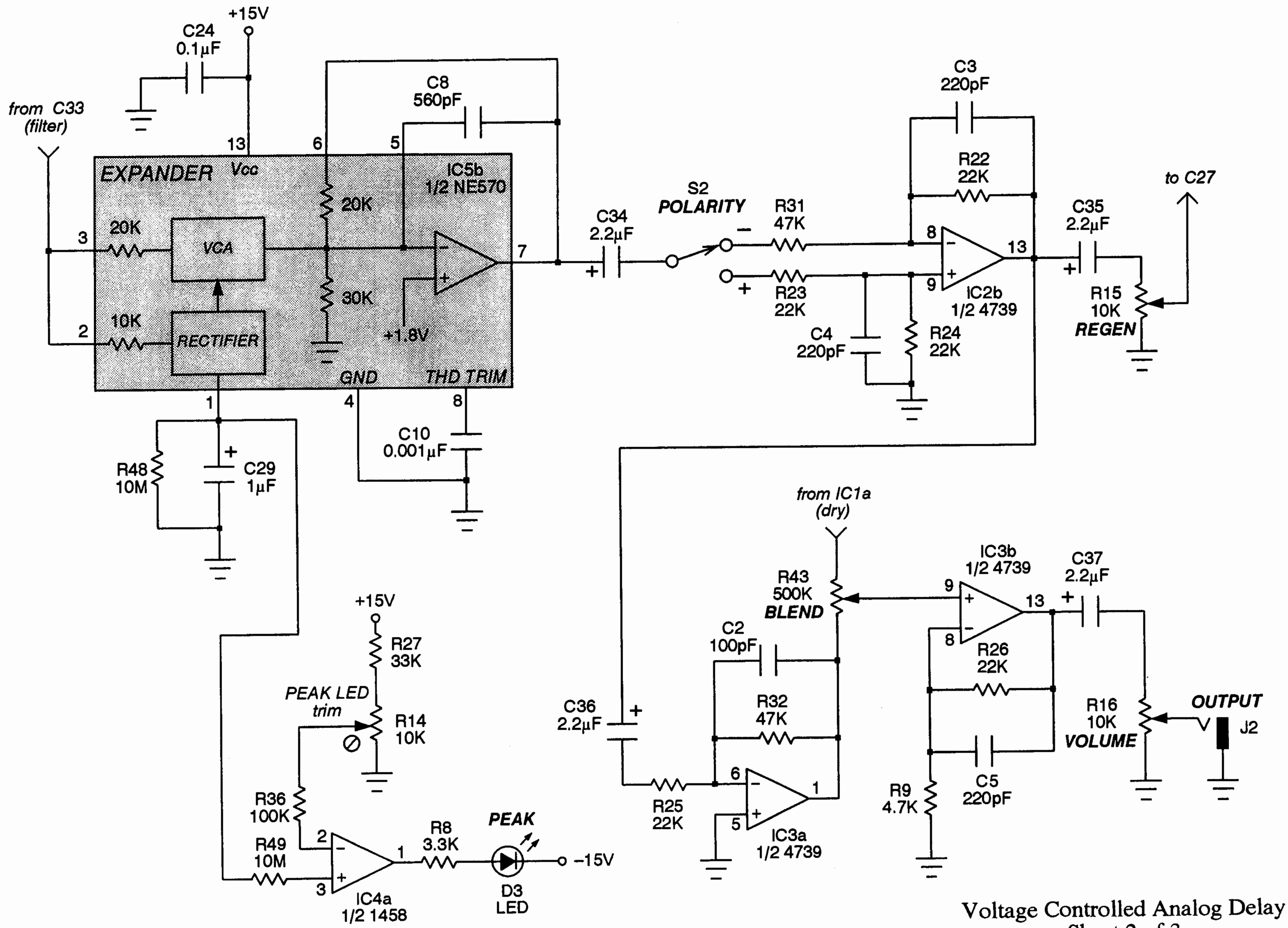


Voltage Controlled Analog Delay
Sheet 1 of 3



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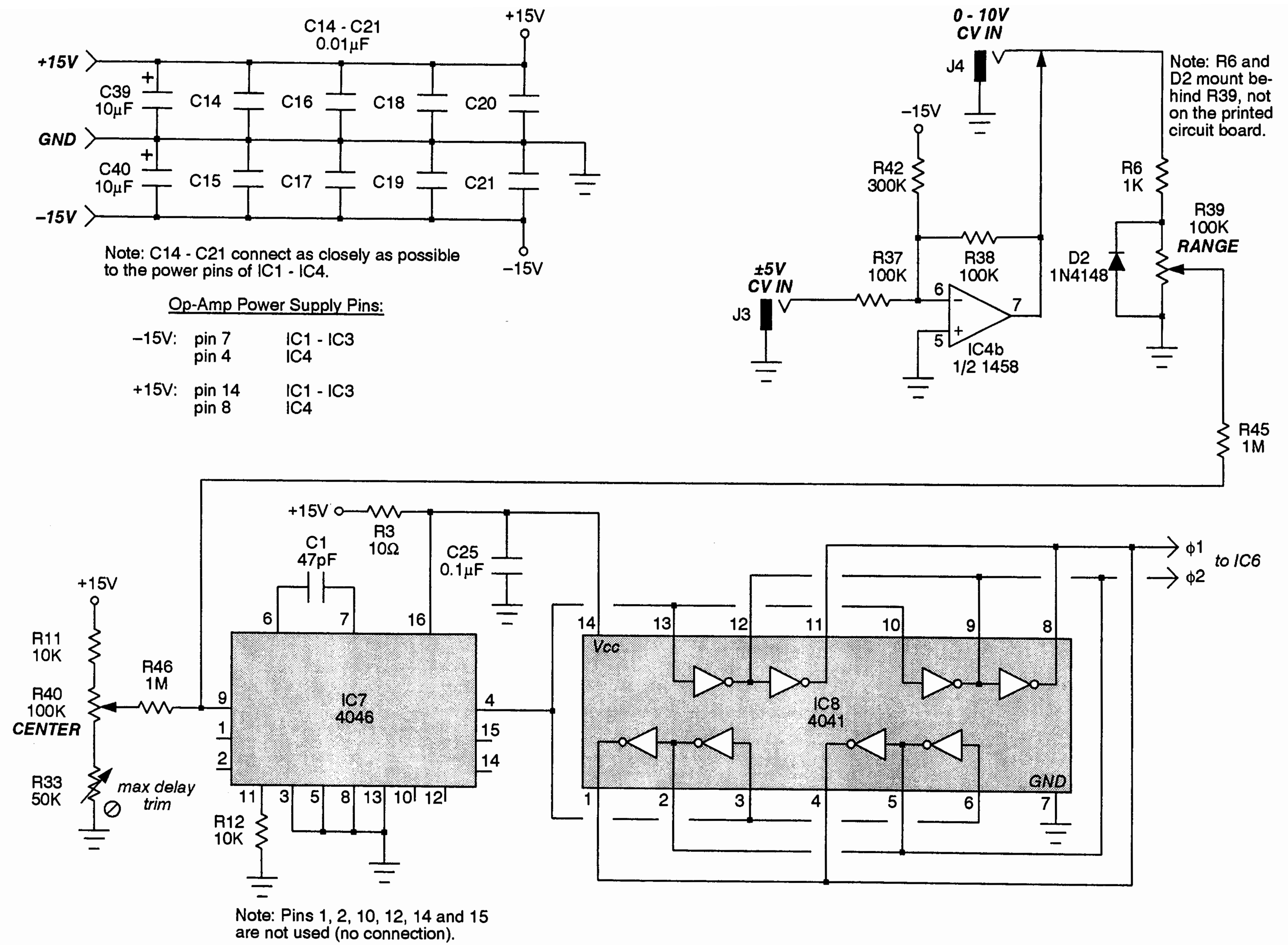
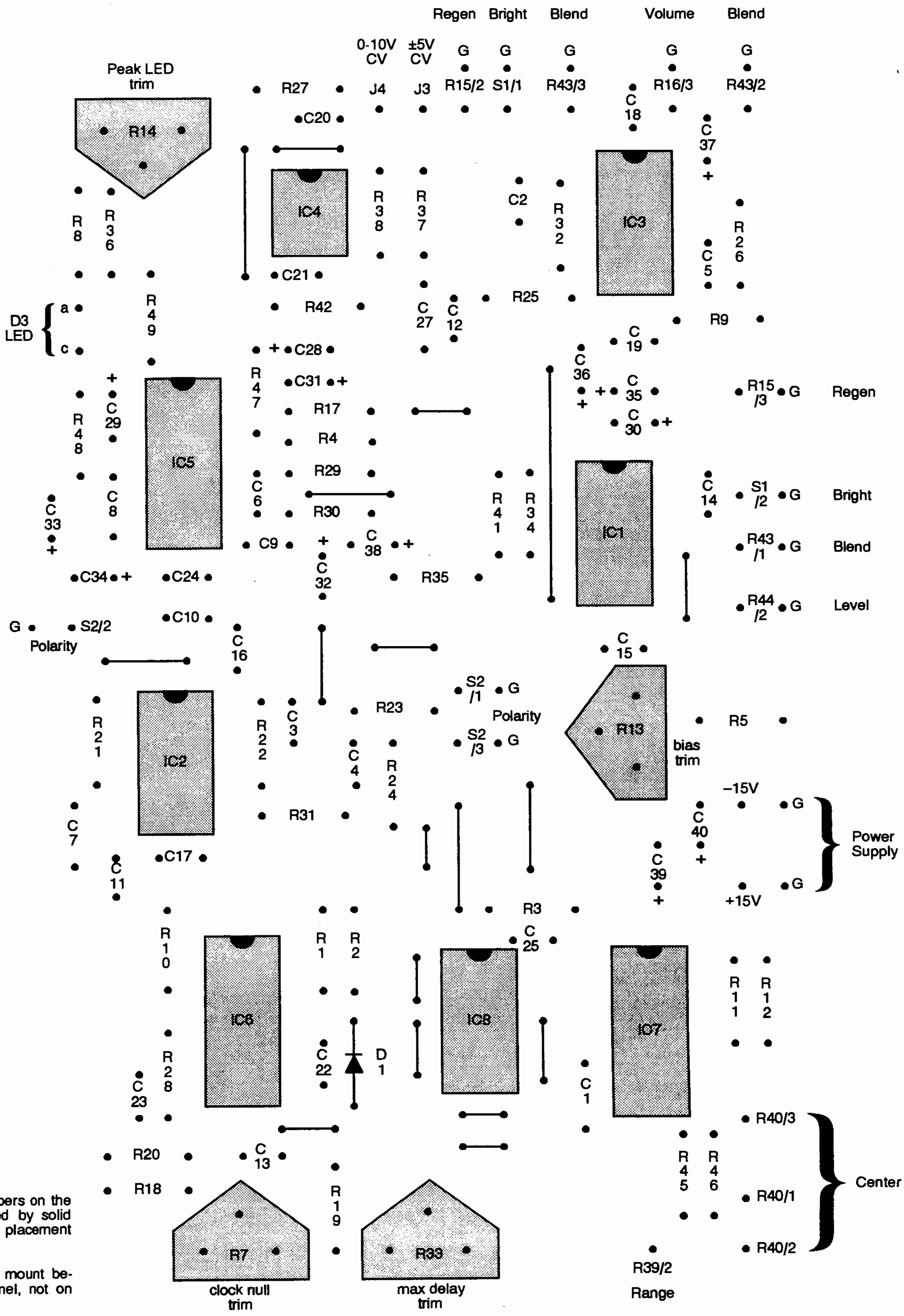


Voltage Controlled Analog Delay
Sheet 3 of 3



Notes:

1. There are 18 jumpers on the board, represented by solid lines on the parts placement guide.
2. R6, D2 and C26 mount behind the front panel, not on this circuit board.
3. Points labeled "G" denote ground. See text for important information on how to ground shielded wires to sensitive audio points.

Parts Placement Guide

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