Craig Anderton

Building the 'Hyperflange + Chorus,' Part I

It has been a while since we've done a construction project, but I think you'll feel it was worth the wait after you see what the Hyperflange + Chorus (H+C for short) can do. The H+C was literally over two years in the making—I've gone through five distinct prototype stages, tweaking each one just a little further, until finalizing the design shown here. Not only has the H+C been extensively tested by me, but PAIA Electronics has been offering it as a kit with excellent response.

What makes the H+C so special? Unlike digital delays which can only sweep over a 4:1 range or so, the H+C typically sweeps from 170 microseconds to at least 15 milliseconds (over an 88:1 sweep range!), which gives a *highly* dramatic flanging effect. Also, compansion helps keep noise levels way down, even at the longer delays which are optimum for chorus effects. Plus, there is a unique modulation section which gives an exceptionally musical sweep effect, choice of positive or negative flanging, voltage controlled inputs, footpedal option...and lots more, as you'll find out as we go along.

I figure that there are two types of people who will build the H+C—those who will buy the parts kit from PAIA, and those hardcore do-it-yourself (DIY) types who will build the thing from scratch. Let's establish at the outset, then, that the H+C is *not* a circuit for beginners. Therefore, this article will not even try to give a blow-by-blow description of the construction—that would take most of the magazine. However, there will be enough information presented for those who are DIY veterans to get this project successfully off the ground.

We will also cover the theory of operation for those of you interested in the workings of delay lines, along with applications. These applications are more-or-less

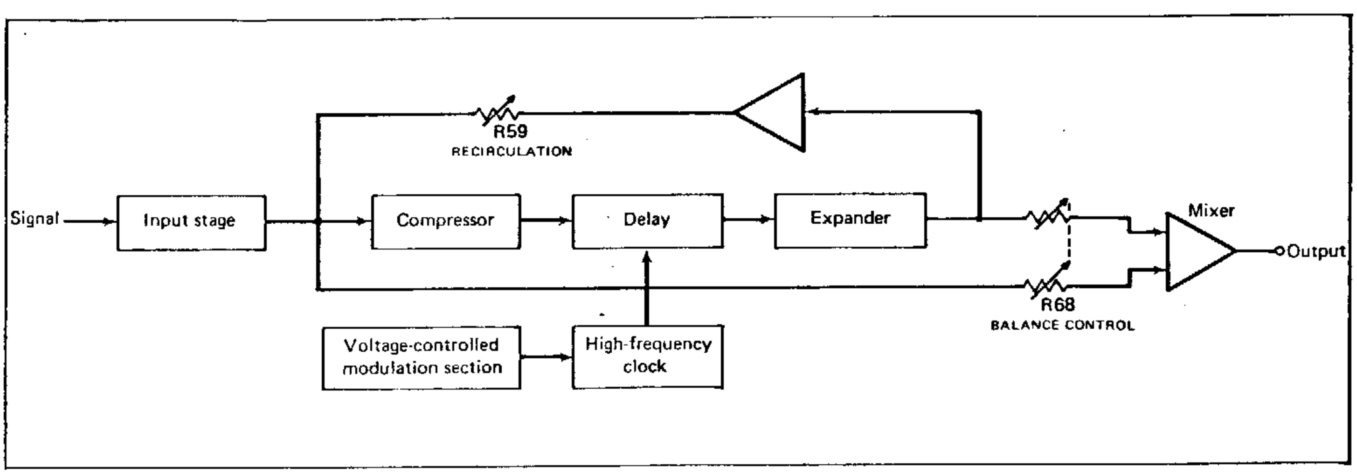


Figure 1. Block diagram of the Hyperflanger + Chorus.

pertinent to other flanging devices as well, so this section will also be of interest to those who don't build their own devices, but would like to get the most out of a flanger or chorus unit which they might already have.

General Information

Before discussing specifics, let's discuss analog delay from a general standpoint.

Different amounts of time delay give different effects. The following are some guidelines as to what kinds of sounds are associated with what kinds of delay times. The delays are so short that we can't describe them in terms of seconds. Instead, we'll be talking about milliseconds (1/1000 of a second), which is abbreviated ms.

0 to 15 ms delays: Mixing a signal delayed by 0 to 15 ms with a non-delayed signal produces flanging sounds. Flanging is a dramatic special effect that imparts a jet airplane-like sound to any instrument or tape track going through the flanger.

10 to 25 ms delays: Mixing a signal delayed by about 10 to 25 ms with a non-delayed signal produces the popular chorusing effect. This creates a fuller, more animated sound that resembles the sounds of two instruments playing at once.

25 to 50 ms delays: This gets into the so-called "slapback" echo range, where you can perceive that the delayed signal is occurring later in time with respect to the non-delayed signal. (With flanging and chorusing effects, it's difficult to tell that an actual delay is taking place because the delay time is so short.)

50 ms and up: This is the range covered by most echo units. Because it is difficult for low-cost analog delay devices to cover this wide a range of time shifting, you'll find that certain devices are optimized for certain ranges. The H+C covers the top two ranges.

How the H+C Works

Figure 1 shows the H+C's general format: an audio signal appearing at the input becomes delayed by a certain amount of time, and this delayed signal appears at the output. The balance control mixes the normal and delayed signals, and the *recirculation* control feeds some of the delayed signal back to the input for "sharper" sounds. A voltage-controlled modulation section lets you vary the amount of delay over at least a 70:1 range, allowing for automatic sweep effects, vibrato, chorusing, stereo simulation, and more.

The heart of this circuit is an analog delay IC called a *bucket brigade device*, or BBD for short. This particular IC includes 1,024 serial stages, each of which is capable of storing a voltage for a short period of time. The BBD samples the input signal at a very fast rate (above the audio range). Sampling is a process whereby the analog delay line stores the instantaneous amplitude value (level) of an input signal in its first stage, while passing previous samples—in a bucket brigade-like fashion—from the first stage down the IC's 1,024 stages. As this process of moving the samples down the delay line progresses, the first sample eventually 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. The actual amount of delay depends upon the sampling rate and the number of stages in the BBD. The sampling rate is controlled by a companion circuit to the delay line known as a clock. The frequency of the clock sets the sampling rate; a slower clock rate transfers samples slowly, while a faster rate moves the samples down the bucket brigade at a faster rate.

Unfortunately, while this all sounds good in theory, some problems creep into the process. First, in order to accurately represent the input signal, we need to take *lots* of samples—maybe one every 50 microseconds (a microsecond equals 1/1,000,000th of a second) or so. This is like the connect-the-dots games that kids play, where more dots improve the resolution of the drawing: more samples improve the resolution (fidelity) of the signal. However, as we increase the sample rate, we're also shifting our signals down the delay line at a faster rate, which gives us less delay. So while we can get a pretty decent sounding output signal with short delay times, at longer delay times our sampling rate goes down and the signal becomes less defined, which we hear as muddiness or distortion. If the clock rate is slow enough to enter the audio range, an annoying whistling tone appears along with the audio.

Increasing the number of stages seems like an easy solution, but, unfortunately, each stage contributes a certain amount of noise, transfer inefficiency, and high-frequency loss. 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.

The H+C Clock

Refer to the schematics *(Figures 2A* and *2B)* as we go through the following sections on theory. (Incidentally, the circled letters on the schematic are keyed to connections on the printed circuit board.)

IC2 is a phase-locked loop set up as a voltage-controlled clock. Varying the voltage at pin 9 varies the clock frequency from 17 kHz to 1.5 MHz, which gives a maximum delay range of 15 milliseconds (0.015 seconds) to 170 microseconds (0.00017 seconds). This covers the flanging and chorusing ranges. R24 sets the initial delay time, while R25 mixes in the desired amount of modulation from the hypertriangular modulation circuit (more on this later).

Since the clock input of a delay line includes some capacitance, which acts like a high-cut filter, at high frequencies you need to deliver lots of current to the analog delay chip in order to charge that clock capacitance as fast as possible. IC3 is a high-current buffer capable of delivering a clean square-wave clock to the delay line, even at high frequencies.

The H+C Modulation Section

While a static clock frequency can produce musically useful results by giving a fixed amount of delay, most musicians prefer to add some modulation to the clock to produce a more animated effect. Delay lines generally use triangle wave LFOs (low-frequency oscillators) which alter the clock rate in a linear fashion. *Figure 3A* shows a typical triangle wave; *Figure 4* graphs an exponential sweep of delay time from longest delay to shortest delay versus clock rate. Note how the first part of the sweep (from 0 to 200 kHz

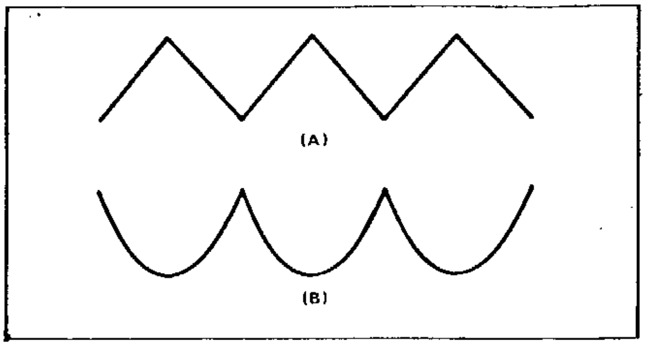


Figure 3. A triangle wave with minimum (A) and maximum (B) "hypertriangularity."

clock rate) covers a great deal of delay and, as the sweep continues, you add less and less additional delay time. The sonic result is that the flanger seems to linger for a long time at the top of the range, then swoops down rapidly into the lower range and almost immediately returns to the top of the range again. Since the lower (longer) delay ranges often create the most interesting sounds, this is not the best way to go. The wider the sweep range of the flanger—and the Hyperflange + Chorus has quite a wide sweep range—the worse a linear sweep sounds.

The first attempt I saw to fix this problem was a clock c[illegible] by Jacques Boileau published in the March/April 1982 issue of *Polyphony* magazine. The "Hypertriangular" clock used in the H+C is a simpler, and more versatile, implementation of his basic idea. *Figure 3B* shows the waveform of a hypertriangular clock. Note that the waveform is exponential in character. This means that the clock sweeps through the upper delay range rapidly and, as it sweeps towards the lower delay range, the clock slows down. The end result is a sweep that sounds like it moves smoothly through the entire delay range, neither spending too much time at the higher end nor spending too little time at the lower end.

We can obtain a hypertriangular sweep waveform by taking a voltage-controlled, low-frequency triangle wave oscillator and feeding its triangle output back into the voltage control input. However, you need an oscillator with excellent stability and wide range. IC1, the Curtis CEM3340, is the perfect (albeit costly) solution. Its triangle output (pin 10) feeds a buffer (IC5) which then drives the voltage controlled input of IC2. Another buffer (IC4) drives the hypertriangularity pot (R28). This pot varies the amount of triangle wave fed back to the voltage control input; maximum hypertriangularity produces the curve shown in *Figure 3B*, while minimum hypertriangularity gives the standard triangle waveform in *Figure 3A* (which is still musically useful for small-range sweeps).

We can vary the modulation speed in two ways: R29 feeds a variable voltage into the CEM3340 to set the initial speed; however, you may further vary the speed (exponentially, even!) by injecting a 0 to +10 V control voltage into J2. For synchro-sonic effects, you can feed sync pulses or square waves into J3; a positive (+10 V) pulse to this jack reverses the LFO sweep direction if the LFO is sweeping upward.

Also note that you can tap the modulation section output via jack J1 if you want to slave two H+C units together, or feed other voltage-controlled devices with the hypertriangular waveform. You may also control the delay by plugging a 0 to +10 V control voltage into J4; however, this interrupts the connection going from the hypertriangular clock to the 4046's control voltage input.

H+C Audio Section

IC8A is a preamp with choice of AC or DC coupling. Use J6 for all applications unless you encounter distortion or other problems; in that case, plug into J5.

The signal leaving IC8A goes through an attenuator pot (R58) which lets the H+C accommodate high signal levels, such as those found in +4 dBm studios.

IC6 is a compressor/expander. The compressor squeezes the dynamic range by a factor of 2:1 by limiting high level peaks and boosting low level valleys. The input signal couples into the compressor in two ways: through an on-chip resistor, and through a pre-emphasis network (C24 and R72), to optionally add more treble to the signal. Capacitor C18 is a high-frequency rolloff capacitor which exhibits variable bandwidth characteristics (i.e. when the compressor's op amp is running at high open loop gain, there is a maximum amount of high frequency attenuation). R70 adds a slight bias to C30, thereby defeating the compression action for low-level input signals. R55 varies the voltage going to the compressor op amp's summing junction, which varies the quiescent output

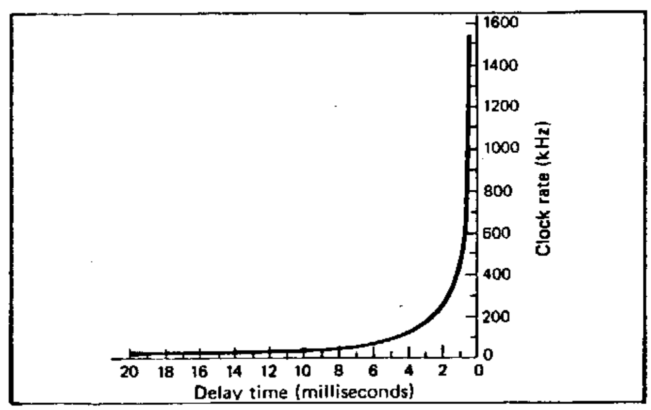


Figure 4. A graph of delay time versus clock rate.

voltage of the op amp and allows direct coupling into IC7. The reason why direct coupling is important is that the input of IC7 wants to see as low an output impedance as possible from the preceeding stage; this minimizes bias variations as the clock sweeps over its full range, which could otherwise restrict signal levels through IC7.

IC7's output, consisting of a series of sampled voltages, doesn't really resemble our input signal since it is more of a "stair-step" waveform than the smooth waveform we had at the input. IC8D is a low-pass filter which not only smooths out the samples, but also minimizes any of the high frequency clock signal that may still be riding along with the audio signal. Most delay lines also use a low-pass filter at the input, to prevent high frequency input signals from interfering with the clock signal; C18 performs this function with IC6.

The filtered signal then goes to the remaining section of IC6, which is hooked up as an expander. This "undoes" the effect of the compressor by adding a

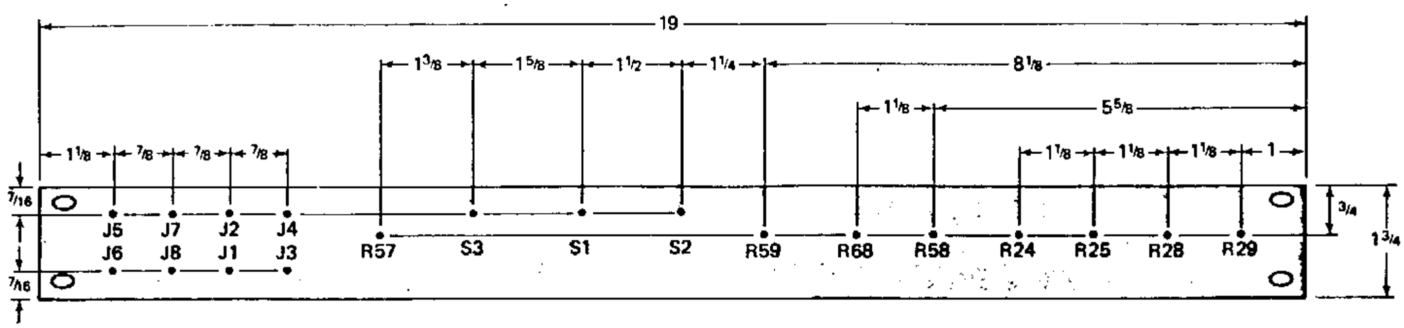


Figure 5. The front-panel drilling template.

complementary amount of expansion (1:2) to the audio signal. Not only does this restore our original fidelity and dynamic range, but, best of all, any noise generated in the delay line section is expanded downward, which greatly improves the signal-to-noise ratio.

Unfortunately, compansion is not perfect—it doesn't eliminate noise, but instead makes it far less prominent. However, when you consider that the alternative is an objectionable noise level, it's no problem putting up with the much less objectionable quirks (occasional pumping or breathing, and an inability to handle the exceptionally fast transients associated with companding circuitry).

IC9 is a clipping indicator that samples the voltage on the expander's filter capacitor (C31). When the voltage on this capacitor exceeds the voltage selected by trimpot R54, LED1 lights to indicate an overload condition.

The expanded output goes in two directions. One is towards IC8C, which recirculates the signal back to the input. S1A alters the phase of this stage to give positive or negative recirculation for two different tonalities. S2 lets you cut the low frequencies to minimize "booming," while R59 determines the amount of recirculated signal.

IC8B combines the delayed and straight outputs to create frequency response cancellations which result in flanging sounds. The non-delayed signal goes directly into IC8B's summing junction (inverting input), while the expander output passes into either the inverting or non-inverting input of IC8B, depending on the setting of S1B. S1B retains phase accuracy of the delayed signal, regardless of whether positive or negative recirculation is being selected. The mixer output then goes through R58, which sets the overall output level, and finally appears at J7 and J8, a pair of paralleled output jacks.

Well, that's quite an explanation! But then again, the H+C is not a simple circuit. Now, let's move on to construction.

Building the H+C

Analog delays involve both audio and radio frequencies. As a result, the circuit board must be very carefully laid out to minimize stray RF. To give you an idea of what I mean, the first prototype I constructed on perfboard had about 2 V peak-to-peak of RF riding along on the ground lines. Proper circuit board layout reduced that by 40 dB (20 mV of RF). While you might think this isn't a problem—after all, you can't hear RF—your tape recorder or other delay lines could very well be affected by stray RF.

For proper operation, you *must* use a properly laid out circuit board. You can either order an etched, drilled, and legended board with component layout, or 1:1 artwork with component layout (if you want to etch your own), from PAIA (see parts list for ordering info). The circuit board artwork is not included with this article because of space considerations (the board is about 11.25" × 3.75").

Once you have the circuit board, install all components and jumpers, and solder in sockets for all the ICs (important!). After mounting all parts except ICs on the circuit board, move along to front panel fabrication.

Figure 5 gives a drilling template for a standard 19" × 1.75" rack panel. The template assumes toggle switches, although you can just as easily use slide or rocker switches with a few minor changes in the layout. *Figure 6* shows a recommended way of mounting the pots, switches, and jacks, viewed from the rear of the panel. The circuit board mounts along the lower edge of the panel.

Once you have the above components mounted on the panel, you need to connect wires between the circuit board and panel-mounted components. For best results, shield the wires going to the balance, input level, regeneration level, and output controls. Connect the circuit board ground connection (pad G10 on the circuit board) to the input-jack ground lug; finally, plug the various ICs into their respective sockets.

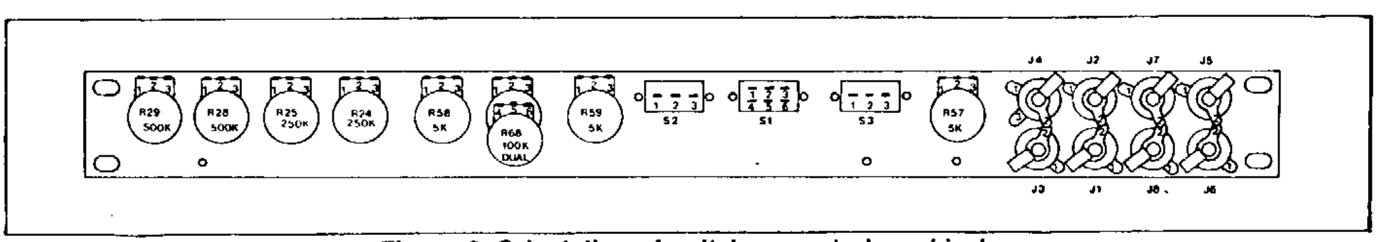


Figure 6. Orientation of switches, controls and jacks, as seen from the rear of the panel.

Building the 'Hyperflange + Chorus,' Part II

Craig Anderton

Installation

You must provide the H + C with a *regulated* ±15 volt power supply or it will not work correctly. Use heavy-gauge wires to connect the supply to the unit, especially if the power leads are more than about a foot long. The most important connection is the ground wire; this should be as thick as possible—even a few thousandths of an ohm can cause a voltage drop which may affect performance.

When using the H + C as a stand-alone unit, simply hook up the power supply +, −, and ground leads, and treat the unit as you would any other effect.

If you decide to mount the unit in a metal rack frame, you can hook up the power connections as described above, and bolt the unit into the mainframe. However, you may experience ground loop problems. If this is the case (as evidenced by hum, whistles, or other strange problems), remove the ground wire running from circuit board pad G10 to the front panel, and connect the rack frame to the power supply ground point through a heavy ground wire. If your rack frame is made of wood, that wire should connect directly to the front panel rather than the rack frame.

In some instances, you might have less problems if you leave the ground wire from pad G10 to the front panel intact. It is impossible to predict which of the above will work best for you; I'd suggest hooking a 'scope probe on to ground and see which way puts the least amount of RF and hum onto the ground line. Improper grounding will not necessarily damage the unit, but it may keep it from working up to its full potential.

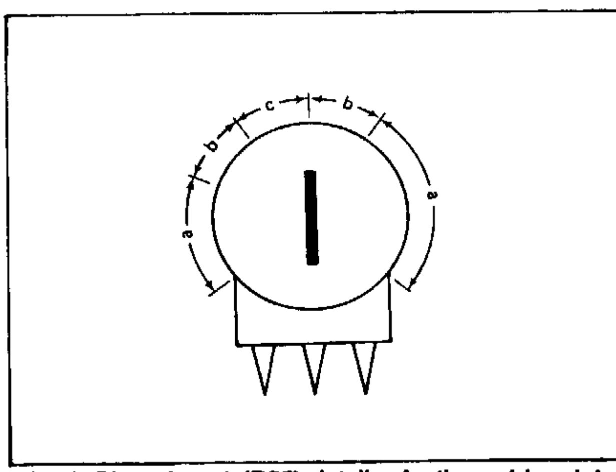


Fig. 1. Bias trimpot (R55) details. As the pot is rotated through regions a-b-c-b-a, you should hear (a) no sound, (b) distorted sound, and (c) clear sound.

Calibration

Correct calibration is absolutely essential for best operation. The following procedure is for those without test equipment; the H + C Assembly and Using manual (see parts list) gives details on calibration using a 'scope or 'scope and function generator.

"CW" means turn a knob clockwise; "CCW" means counter-clockwise. Positions in between these extremes are indicated in the same way as a traditional analog clock—for example, 12:00 means the knob pointer should point straight up.

Start off calibration with the knobs and switches set as follows: Input level, 12:00; Pre-emphasis, off; Regeneration phase, (+); Lo cut, off; Regeneration level, CCW; Balance, CCW; Output, 12:00; Initial delay, CCW; Modulation depth, CCW; Hypertriangularity, CW; LFO rate, 12:00; all trimpots (except R54), CCW as viewed from the front of the trimpot. Set R54 CW.

Begin calibration by hooking up a regulated ±15 volt power supply, plugging a signal into AC coupled input jack (J5), and listening to the output through an amplifier. Since the balance control is selecting dry signal only, you should hear the unmodified sound of your instrument. If not, check over your wiring, power supply, input signal and amp. Otherwise, proceed as follows.

Turn the Balance control CW and turn your amp level down to prepare for a LOUD sound. Adjust Low Clock Trim R16 CW for the highest pitched audio tone you can still hear. Now, adjust Clock Cancel trim R56 until you hear the minimum amount of high frequency tone; this should occur towards the middle of the trimpot. As you reach this "null point," you will need to turn up the amp level to hear the signal better. You should be able to almost completely eliminate this tone through careful adjustment of R56.

Plug an instrument or other signal source into the input, and turn the Initial Delay control to 9:00. You shouldn't hear anything as you play. Now it's time to adjust the bias; this is a critical adjustment, as lowest distortion, greatest dynamic range, and lowest noise all depend on the proper calibration of this control.

As you play your instrument, vary R55, the Bias trimpot. At some point in its rotation, you'll hear your instrument briefly, after which the sound may go away. So far, so good. Now for the fine adjustment.

Vary the trimpot in *small* increments. Referring to *Figure 1*, as you vary the trimmer from CCW to CW you'll go through a band of no sound, then lots of distortion, a clear spot where the sound is undistorted, more distortion, and then another band of no sound. *Figure 1* is approximate—your trimpot may have these sounds occur at somewhat different places, but the basic drawing is close enough. Remember to vary the trimpot only a little bit at a time, since it takes some time for the adjustment to "settle" whenever you make a change.

Continue by setting R55 for minimum distortion. If there is no spot which is undistorted, then you are feeding in a fairly high-level signal. Adjust R55 for as little distortion as possible, then pull back a bit on the Input Level control until the sound clears up.

Now that you have a clean sounding signal, increase the input level until you hear just a hint of distortion. Readjust R55 in small increments until the distortion goes away. (Throughout these procedures, if you are using an instrument [such as a guitar], play in a consistent manner to make calibration easier.) Increase the level some more, and again adjust R55 for minimum distortion. Past a certain point, R55 will no longer be able to trim out distortion; when this happens, back off a bit on the input level, adjust R55 for minimum distortion again (if necessary), and the bias will be properly calibrated. To sum up: The object of this exercise is to have the *lowest* possible amount of distortion with the *highest* possible input signal level.

To calibrate the modulation section, set the LFO rate CW, Hypertriangularity CW, and Modulation Depth CW. Adjust R27 in a clockwise direction; at some point, you'll hear a periodic modulation of the signal which is similar to vibrato. Turn R27 CCW until the modulation just disappears, then CW until the modulation just kicks in. Vary the Hypertriangularity control; note that as you vary it from CW (hypertriangular) to CCW (normal triangle), the LFO speed increases somewhat.

Finally, play your instrument and turn up the Input Level control so that you hear distortion on peaks. Adjust clipping trimpot R54 so that the overload light flashes whenever the distortion occurs.

Whew... that took some work, but the unit is finally built and calibrated! Now let's take a general look at what the jacks and controls do, and finally, cover applications.

Jacks

IN (AC). This is a capacitively coupled input to the H + C. Please note that AC does not refer to AC line voltages, but to the nature of the electrical coupling used. Do *not* under any circumstances plug an AC cord into this jack, or you will seriously and irreversibly damage the unit.

IN (DC). This input couples directly into the Hyperflange without any coupling capacitor. It is the recommended input for electric guitars and any other equipment that does not exhibit a DC offset. The input impedance is 470k, so you can plug a guitar in without suffering any loading.

OUT 1, 2. Two paralleled output jacks. Note that these are not synthesized stereo outputs, but rather, two identical outputs.

LFO CV IN. Applying a 0 to +10 V control voltage to this jack alters the speed of the LFO in an exponential fashion.

LFO OUT. Taps off the LFO. Provides a hypertriangular or standard triangular signal to other voltage-controllable units, or can slave two H + C units together. (For how to slave one unit to another, see the section on Applications.)

SYNC. A positive +10 V pulse to this jack reverses the LFO sweep direction if the LFO is sweeping upward.

DELAY CV IN. Applying a 0 to +10 V control voltage to this jack linearly varies the frequency of the clock controlling the delay time from maximum to minimum, while disconnecting the internal LFO from the clock. The initial delay and modulation depth controls are still active in this mode.

Controls

INPUT LEVEL. Turn up for lower level signals; turn down for higher level signals. Adjust for maximum level short of the overload indicator flashing.

PRE-EMPHASIS. When you want to add a trebly "zing" to your signal, switch this to "on." However, attempting to add pre-emphasis with a signal that has lots of high frequencies could overload the circuitry as well as cause serious "aliasing" (whistles etc.). You will not be able to add pre-emphasis under all conditions, but in most applications it will work just fine.

REGENERATION PHASE/LEVEL. The (-) position gives a whooshing, breath-like sound, while the (+) position imparts a more metallic, zinging sound. Note that it is impossible to turn up the Regeneration Level control to the point where the circuit breaks into loud oscillation. To lock the resonance below a certain level, turn the Regeneration Level up all the way and play your instrument; you should hear a loud tone. Keep playing, and turn the Regeneration Level CCW just until the oscillation stops. Unless there is a significant increase in input level, the unit will not break into oscillation upredictably.

LO CUT. In the ON position, this switch introduces a low-cut filter in the regeneration path that prevents low frequency resonance. In the OFF position, all frequencies are regenerated with equal intensity.

BALANCE. CCW gives straight sound only, CW delayed sound only. In the mid-position, there is an equal balance of delayed and straight sounds.

INITIAL DELAY. CW gives minimum delay, CCW maximum delay.

MODULATION DEPTH. This control determines how much the delay line is affected by the LFO sweep. CW gives the widest sweep range; CCW gives no sweep at all.

HYPERTRIANGULARITY. Full CW gives maximum hypertriangularity; CCW gives a standard triangle wave.

LFO RATE. CCW gives the slowest sweep, CW the fastest.

APPLICATIONS

Now comes your reward for slogging through all the technical stuff: you get to create some really intriguing sounds. In the following examples, Input Level, Pre-emphasis, Regeneration Lo Cut (when regeneration is specified), and Output Level are adjusted to suit. All indicated control settings are initial settings, and are intended to provide a point of departure for further experimentation.

Chorusing

This gives a lush, full sound. The crucial controls are Initial Delay, Modulation Depth, and LFO Rate. If the Initial Delay is too long, then the sound will resemble slapback echo more than chorusing; but if it's too short, then you'll hear more flanging-like sounds. Modulation Depth should be moderate, since at longer delays too much depth can give unpleasant detuning effects (as well as make the chorus sound less interesting). Should detuning effects occur, slowing down the LFO Rate will usually minimize any problems. Triangle waveforms work well for chorusing at fast LFO speeds.

Regen Phase	Regen Level	Bal-anace	Init Delay	Mod Depth	Hypertri	LFO rate
(+)	9:00	12:00	9:00	9:00	12:00	12:00

Auto Sweep Flanging

With this patch, Initial Delay sets the bottom of the flange sweep, while Mod Depth sets the top of the sweep. Setting the Initial Delay for the longest possible delay can give you some dramatic effects with slow LFO rates. If you want a vivid demonstration of why hypertriangularity is needed, try this same patch but turn the Hypertriangularity control full CCW so that the flanger is swept with the standard triangle wave. You'll find the difference *very* convincing!

Regen Phase	Regen Level	Bal-ance	Init Delay	Mod Depth	Hypertri	LFO rate
(+)	2:00	12:00	9:00	CW	CW	12:00

Pedal Flanging

Same as above, but plug 0 to + 10 V control voltage pedal output into the DELAY CV IN jack.

Vibrato

Critical controls are Hypertriangularity (which determines the "smoothness" of the vibrato) and LFO Rate. Incidentally, for foot pedal controlled vibrato (where pushing down the pedal injects vibrato), patch the LFO OUT jack into the instrument input of a pedal, and then patch the amp output of the pedal to the DELAY CV IN jack. This lets guitarists vibrato entire chords—not just single notes—under foot control.

Regen Phase	Regen Level	Bal-ance	Init Delay	Mod depth	Hypertri	LFO rate
(-)	CCW	CW	1:00	12:00	1:00	4:00

Fast Rotating Speaker Sound

This is similar to the above patch. When simulating slower rotating speakers, you will probably have to pull back on the Modulation Depth a bit.

Regen Phase	Regen Level	Bal-ance	Init Delay	Mod Depth	Hypertri	LFO rate
(-)	CCW	12:00	12:00	12:00	1:00	4:00

Phase Shifter Simulation

While flangers and phasers are quite different, this patch does a reasonably good phase shifter imitation. Be careful not to set the Initial Delay to full CCW, since phasers don't give much of a time delay. For the most realistic sound, set all controls more subtly than you would for flanging effects.

Regen Phase	Regen Level	Bal-ance	Init Delay	Mod Depth	Hypertri	LFO rate
(-)	10:00	12:00	12:00	3:00	3:00	12:00

Notch Filter

Not all flanger sounds involve the LFO. For static, equalizer-like effects, vary the Initial delay and Regeneration Level controls for the desired tonality.

Regen Phase	Regen Level	Bal-ance	Init Delay	Mod Depth	Hypertri	LFO rate
(-)	vary	12:00	vary	CCW	CCW	CCW

Stereo Simulation Spreading

Set up the flanger as follows: Split the signal coming from your instrument or tape track. Take one split and feed that output into one channel of an amp or mixer. Take the other split and patch it into the Hyperflange + Chorus, and patch the H + C's output into the second channel of an amp or mixer. Adjusting the Initial Delay alters the separation effect between the two channels. If you want some "motion" to this simulated stereo, simply add some Modulation Depth.

Note that combining two "stereo" channels created in this manner back into mono will usually affect the timbre of the signal.

Regen Phase	Regen Level	Bal-ance	Init Delay	Mod Depth	Hypertri	LFO rate
(-)	CCW	CW	vary	CCW	CCW	CCW

Slaving Two Units Together

Splitting a signal through two H + C units, with each output going to a different channel, can provide dramatic stereo effects (particularly if the settings on the two units are somewhat different). If you want them to both track the same LFO, run a patch cord from the LFO OUT jack of the first unit to the DELAY CV IN jack of the first unit to the DELAY CV IN jack of the second unit. The second unit will now be slaved to the first unit.

Closing Comments

This is the most complex do-it-yourself project I've presented to date in any magazine, but I feel the performance will really turn on *MR&M* readers—especially considering the cost. The parts list gives you options on materials available from PAIA to help you build the unit—from circuit board artwork to complete kits. Take your time, be patient, follow instructions, and you'll have a flanger which will delight you with its cost-effectiveness.

Specifications: Delay Channel

Dynamic range, unweighted: 80 dB
Dynamic range, "A" weighted: 86 dB
Maximum input before clipping, any delay time, Input Level up full: 2 V p-p
Frequency response, R42 and R43 =22k: -6 dB, 10 Hz; -2 dB, 20 Hz; 0 dB, 50 Hz; +2 dB, 4500 Hz; -6 dB, 8500 Hz; -20 dB, 12 kHz. With R42 and R43 = 10 k, response is flat to 10 kHz. However, this isn't recommended—the noise becomes more obvious, but there isn't a whole lot of musical energy up there worth flanging.
Pre-emphasis boost: +14 dB@ 6 kHz (excellent for guitar and voice)

Other Specifications

Straight channel frequency response: + 1 dB, 20 Hz - 20 kHz
Maximum notch depths: 60 dB
Sweep range, manual control: 15 ms to 250 μsec (60:1)
Typical sweep range, LFO control: 15 ms to 170 μsec (greater than 88:1)
Maximum output @ 1 kHz into 1 k load: 20 V p-p

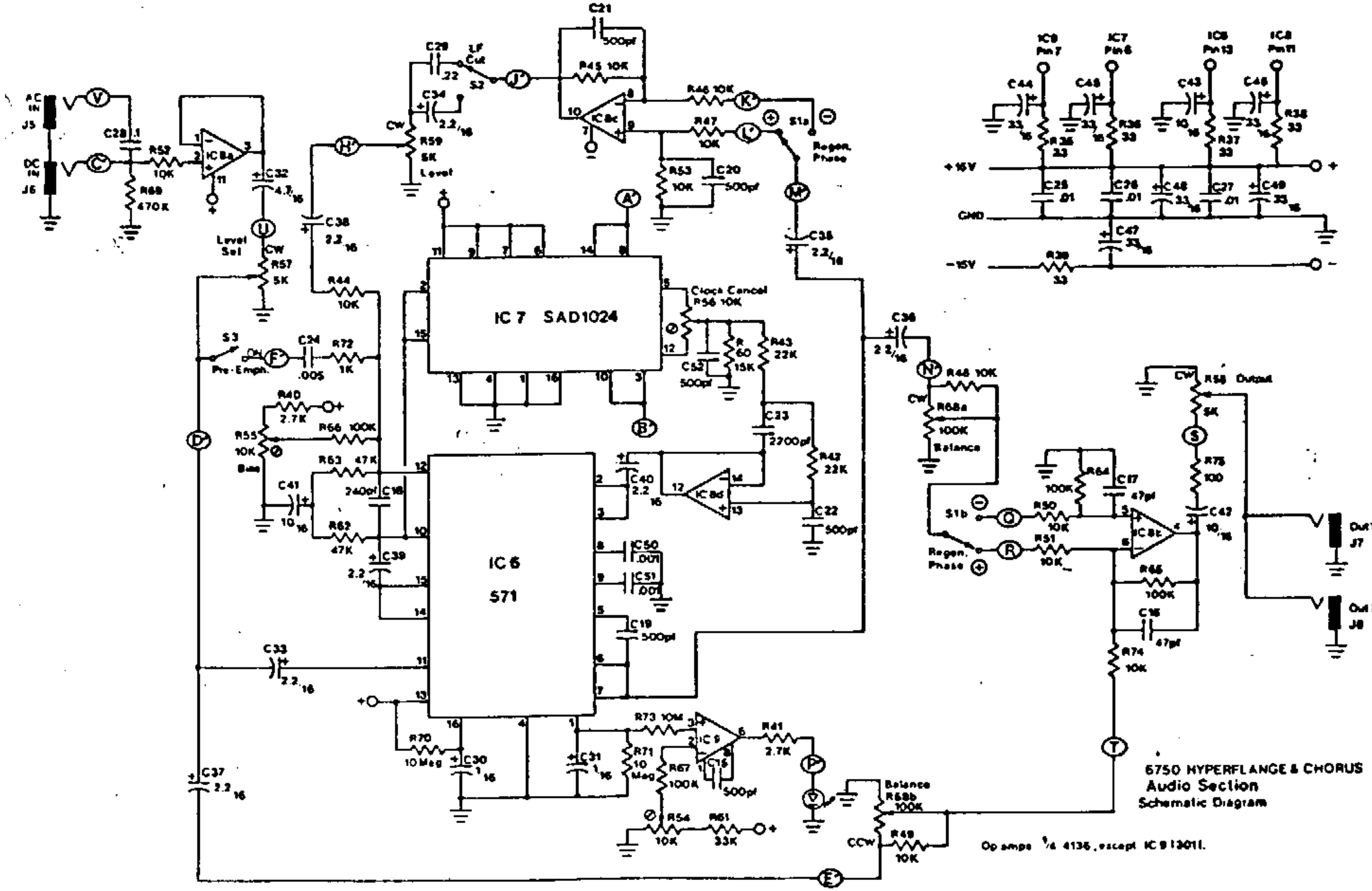
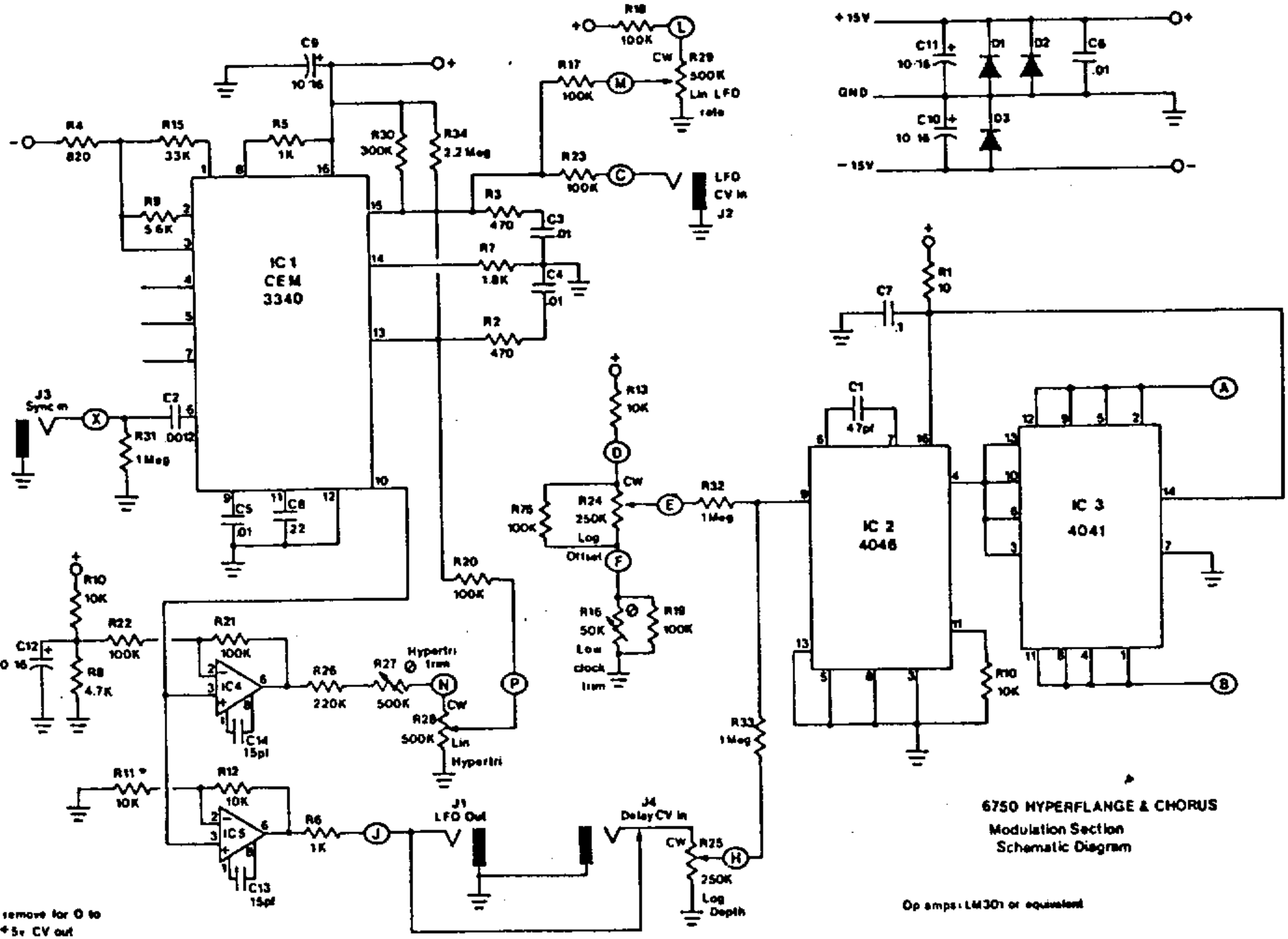


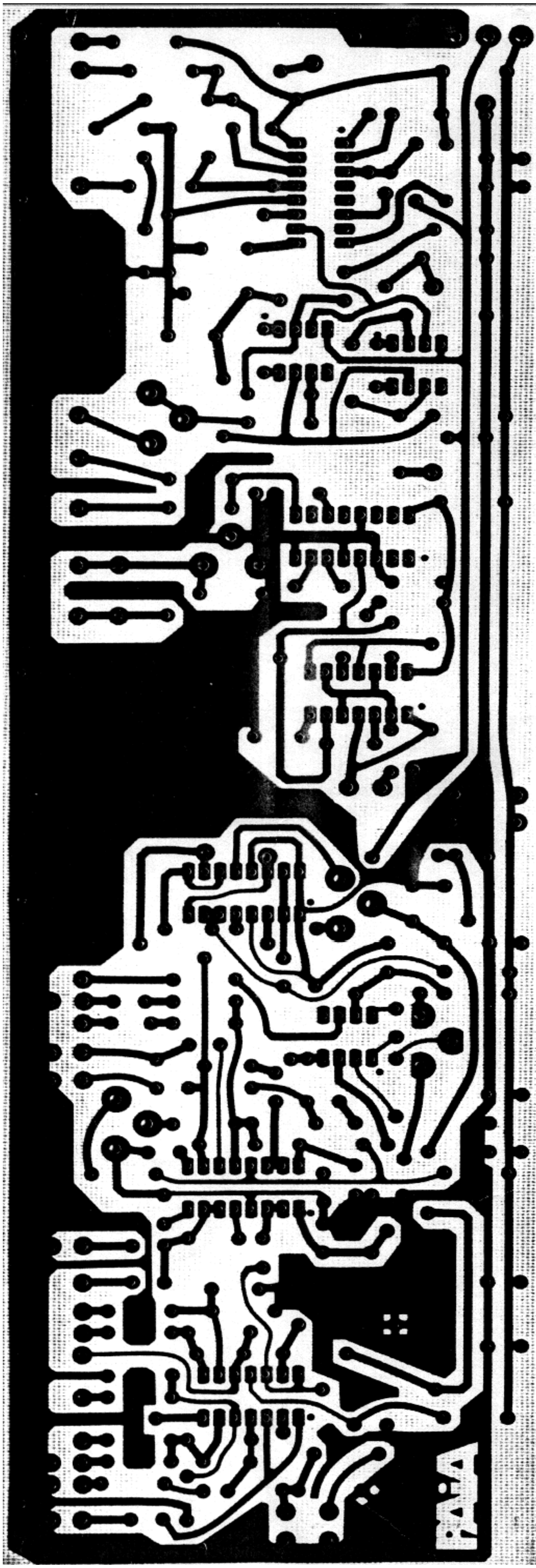
Figure 2. Schematic diagrams for (A) the modulation section, and (B) the audio section. Electrolytic capacitors are labelled with capacitance/voltage.

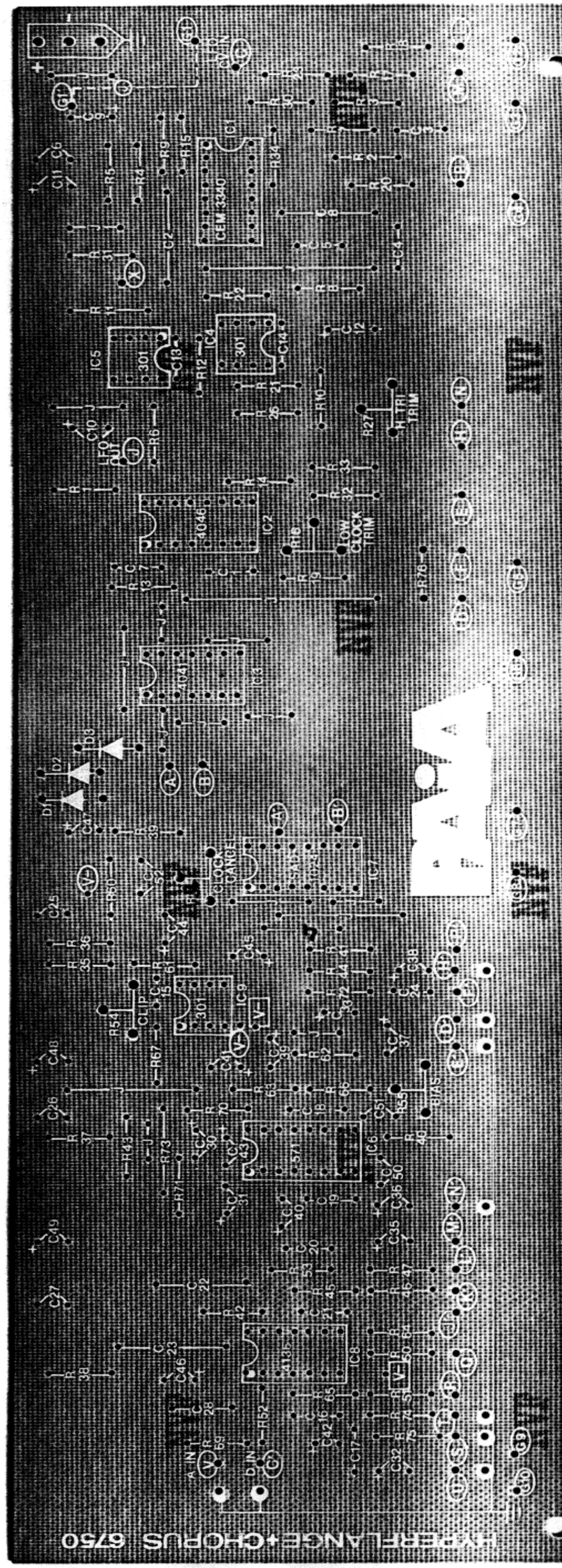


* remove for 0 to +5v CV out

6750 HYPERFLANGE & CHORUS
Modulation Section
Schematic Diagram

Op amps: LM301 or equivalent




HYPERFLANGE+CHORUS 6750