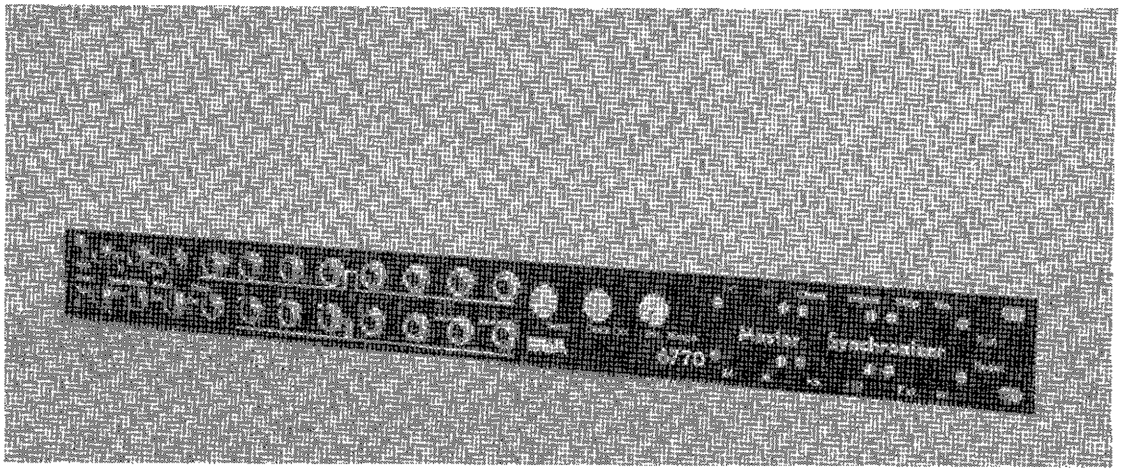
PAIA MASTER SYNCHRONIZER 6770



INTRODUCTION

Thank you for buying the PAIA 6770 Master Synchronizer kit. We realize that you are anxious to get on with the assembly, but before you start, please take the time to read the hints and suggestions that follow.

First of all, this is not a typical PAIA assembly instruction manual. It is assumed that anyone assembling this particular kit is familiar with assembly and soldering techniques. You'll notice that there are none of the usual step-by-step instructions for installing components on the circuit board or doing point to point wiring, so if you are unsure about your ability to complete the assembly successfully, DO NOT BEGIN ASSEMBLY. Instead, see about getting an experienced friend or paying a pro to assemble the kit for you. Meanwhile keep the packaged kit together and intact in case you wish to return it.

The first few pages of this manual comprise a reprint of Craig Anderton's article on SYNTHESIS & SYNCHRONIZATION from Keyboard magazine, January/February 1983. If you have not read this atricle, it will be absolutely necessary to do so. If you are already familiar with the article it can't hurt to go over it again.

Turning Craig's Master Synchronizer into a PAIA kit has generated a couple of small technical discrepancies between the unit described in the accompanying article and the PAIA kit.These are as follows;

1.The INITial TEMPO control, R38, has been changed from the 100K reverse audio tapered pot to a 250K audio tapered pot. The 250K pot supplied in the kit is not wired the same as the 100K pot. The correct wiring connections are shown in the drawings comprising figure B in the back of the manual.

2.Three of the light emitting diodes, D19, D20, and D21, have been replaced by one 5.6 volt Zener diode, designated as D19 in the PAIA schematic and on the 6770 circuit board.

The above changes, as well as differences in IC pin numbers and schematic locations of wire connection points are shown in the PAIA 6770 schematic drawing, (figure C in the back of the manual).

The PC board does not mount on the front panel in the usual bolt-on manner. It is mounted by soldering the HOT lugs of the BOTTOM row of jacks (From Tape, Metronome Out, etc.), to respective solder pads on the circuit board. Be sure to tin all 11 solder pads and jack lugs before mounting the board, and leave the jacks slightly loose in thier holes until all the solder connections are made. Then tighten them carefully so that they don't twist and rip the conducting foil from the circuit board.

Use the bare wire supplied with the kit to make the 23 wire jumpers on the circuit board (designated by solid lines broken by the letter "J" printed on the top of the circuit board), and to make the ground connections on the front pannel. The supplied insulated wire should be used for all other connections between the front panel and the circuit board.

It is far easier to install the panel mounted LED's and make their connections to the circuit board BEFORE making the board to panel wiring connections. Dress the insulated wires in a neat bundle along the length of the panel and in the center between the two rows of jacks.

BEFORE BEGINNING ASSEMBLY, carefully check the supplied parts against the PAIA 6770 parts list in the back of the manual. Check to be sure that all the IC's are in the IC pack BEFORE you open the pack. If any of the parts appear to be missing or damaged, STOP! Contact PAIA Electronics Inc. 1020 W. Wilshire Blvd. Okla. City, OK. 73116. Phone (405) 843-9626.

Finally, if you experience any technical difficulty with your kit, write to PAIA Technical Service (address as above), or phone 1-405-843-6435 9AM to 5PM CST Monday, Wednesday, or Friday.

Craig Anderton's Electronic Projects, No. 8

SYNTHESIS & SYNCHRONIZATION

Interfacing Keyboards, Sequencers, & Drum Machines

ONE OF THE MAJOR DIFFERENCES between acoustic and electronic instruments is that the latter may be linked together and synchronized electronically. For example, a programmable drum unit may send out timing pulses to a sequencer set to generate bass lines, thus insuring that the bass and drums are precisely in sync.

This notion bothers some purists, and understandably so — there are already many instances where people have ceded their humanity to machines. However, the human brain can only handle so many things at once, and if rhythm is taken care of electronically, that frees the player to concentrate on other aspects of playing, such as timbre. Consider the various "electronic" bands currently enjoying popularity in Europe and to a certain extent in the United States, such as Human League, Devo, Soft Cell, Depeche Mode, *et al* (see the June '82 issue of *Keyboard* for an extensive article on the new wave of electronic bands). Many of these bands include members who freely admit that they don't have great "chops" and have only been playing synthesizers for a few years. Yet they are capable of making music which they and the public enjoy; and after all, isn't that what music is all about? — the opportunity for like-minded people to get together and make, or listen to, some pleasing sounds. Of course, you can't help but respect and admire someone who studies an instrument intensively for years, developing flawless technique; yet unfortunately, the technique sometimes becomes an end in itself, and in that case the human quality of the music suffers just as surely as if the player had turned his or her soul over to a machine.

Several years ago I coined the term "synchro-sonic" for music which relies heavily on electronic synchronization. Interestingly, synchro-sonic music produces a couple of effects that I don't think most people would anticipate. One of these is that it seems to take fewer instruments to make a big sound when they are all synchronized. With some of my own music, people have commented about the density of sound, without really noticing that there may have only been three of four instruments playing at one time. I feel the reason for this is that if you are processing an instrument such as keyboard, bass, or guitar in a percussive manner, then that instrument performs double duty, being percussive (rhythmic) as well as melodic. Another effect is that drum parts need not be as complex, since the other instruments are helping hold down the percussive end of things. This is all for the better, considering that electronic drum units lack much of the dynamics and complexity that a human drummer using traps can create.

Behind all synchro-sonic music is electronic music technique and technology, which brings us to the purpose of this two-part article. Many existing devices already have synchronization inputs and outputs which allow different instruments to "talk" to each other, yet musicians are often unclear as to how to use these connections. Worse yet, there is no real industry standardization with regard to sync inputs, so while you may be able to trigger keyboard A with drum unit B without any problems, trying to get that same drum unit to sync up with keyboard C might be much more difficult.

Craig Anderton wears many hats: He is a highly respected expert in musical electronics, the author of two books, Electronic Projects For Musicians *and* Home Recording For Musicians, *the editor of* Polyphony *magazine, a longtime columnist for* Guitar Player, *and a former columnist for* Keyboard *as well. His most recent articles for* Keyboard *were "Delay Lines" (Aug. '82), and "Signal Processing Chains" (Oct. '82).*

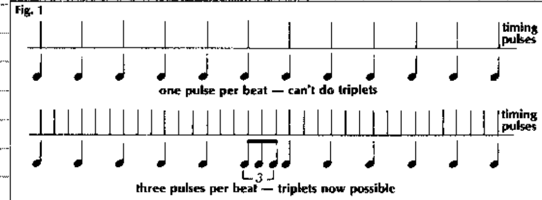
In Part 1 of this article, we'll discuss the basic principles of synchronization, and how various manufacturers approach synchronization. Then we'll discuss ways to achieve synchronization effects using common synthesizer modules — even if you don't have specific sync inputs on your equipment.

In Part 2, which will appear next month, we'll present construction plans for the "Master Synchronizer," a synchro-sonic device intended for both studio and live performance. In the studio, this module can dump timing information on tape and retrieve it — sort of like a glorified click track. It not only allows you to sync up many sequencers and drum machines when doing overdubs, it can also drive some novel synchro-sonic control voltage generators which are inexpensive and *very* useful musically. Features of the Master Synchronizer include a voltage-controlled master clock, suitability for use with virtually any kind of tape deck from cassette to multi-track, audible and visual metronome, and individual outputs with frequency dividers which let you select various sub-multiples of the master clock. We'll also talk about some useful accessories you can build to enhance the unit's effectiveness. Because of its low cost, there are some limitations; but I've used this system in my own studio for about four years with excellent results, and I think you'll enjoy it too.

Basics Of Synchronization. There are two basic types of synchronization, which we'll call "direct sync" and "divided sync." With direct sync, sending a pulse to a unit directly triggers an event; for example, most simple sequencers can be triggered by some kind of pulse which advances the sequencer one step per pulse. Another example would be a synthesizer which has the capability to trigger an envelope when fed with a trigger pulse of some kind.

With divided sync, the master clock operates at a much higher frequency — typically 24, 48, or even 96 pulses per beat in commercial equipment. The reason for using this high frequency is that it gives you much greater resolution. For instance, in the example given above of direct sync, what if you wanted to advance a sequencer at the rate of one beat per sync pulse, but then wanted to play a triplet? With direct sync, you couldn't. However, if you multiply the clock pulse feeding the sequencer by 3, then you're in business: You simply select every third pulse when you want to advance the sequencer on each beat, and when you want to do the triplet, feed each pulse in to the sequencer (see Fig. 1).

Fig. 1



Clearly, the higher the frequency, the greater the resolution. For example, with a 48-pulses-per-beat clock (as used with the Linn drum machine), you could divide a beat up into 48 discrete segments — plenty of resolution for even the fastest runs, or for creating timing patterns which slightly lead or lag the beat.

Synchronization In Commercially Available Equipment. Some manufacturers make families of equipment which are specifically designed to sync up with each other; in fact, the availability of this kind of equipment seems to be on the rise. One of the most encouraging signs that I've heard about is an attempt by some manufacturers to create a standard Musical Instrument Digital Interface (MIDI). A major aspect of this effort is to create a standard number of pulses per beat, so that any equipment adhering to this standard can mate with other equipment adhering to the standard.

Let's look at some representative equipment designed for synchronization. Note that some of these are more or less self-contained "systems" and use divided sync (although they can usually drive direct sync units as well), while others are only capable of direct sync.

One of the most ambitious self-contained sync setups is "The System" from Oberheim Electronics, which comprises the OB-Xa or OB-SX synthesizer, the DMX programmable drum unit, and the DSX sequencer. All of these are intended to work together as a system based on divided sync; however, the DSX also outputs eight external control voltage and gate outputs, which may drive other synthesis gear. The DSX accepts 96 pulses per beat, while the DMX internally divides a 96-pulses-per-beat signal by two to work on a 48-pulses-per-beat standard. (By the way, the Olivia Newton-John song "Heart Attack" makes extensive use of the Oberheim System.)

Roland has been quite active in the synchronized instrument field lately — in fact, the TR-606 Drumatix and TB-303 Bass Line sequencer (reviewed by Dominic Milano in the Sept. '82 issue of *Keyboard*) are specifically designed to sync up with each other. Like all recent Roland equipment, they work off a 24 pulses per beat standard. Since this is an exact sub-multiple of the Linn and Oberheim drum machines, the TR-606 and TB-303 may be triggered from pulses given by these non-Roland drum units.

The Roland MC-4 Microcomposer is perhaps the most universal of all the equipment I've seen with respect to timing. Via its keypad, you can enter any number of pulses per beat, from 0 to 2000. Thus, to interface it with an Oberheim or Linn drum unit, you would simply tell it to work with 48 pulses per beat.

Roland also makes some products which have direct sync capabilities. For example, the JP-8 polyphonic synthesizer has an automatic retriggering input which accepts an input pulse. By deriving a pulse from a drum unit or other source, you may trigger the JP-8's envelope automatically for a dugga-dugga-dugga-dugga effect. If that pulse occurs on the beat, the retriggering will also occur on the beat. If the pulse is a series of sixteenth-notes, then the retriggering will occur four times for each beat. Remember that when using direct sync in this manner, the envelope generator parameters must be carefully set — if, for example, the decay time is longer than the pulse period, the envelope generator will appear to never shut off.

Korg makes several syncable products. Their latest drum unit, the KPR-77, is compatible with the Roland TR-606 Drumatix and TB-303 Bass Line sequencer. The KR-55 drum unit is well suited to direct sync applications, since it has a trigger output which optionally occurs every sixteenth-note, eighth-note, or quarter-note, or may instead follow the bass drum. If you had a JP-8 synthesizer interfaced to the KR-55, for example, you could retrigger the JP-8 at regular intervals or in tandem with the bass drum. The KR-33, a low-priced rhythm unit, allows the option of putting out a trigger simultaneously with the bass drum signal.

With respect to Korg keyboards, both the Mono/Poly and the Polysix include sync inputs for their arpeggiators. The Trident (models I and II) includes a sync input which allows you to retrigger the brass section filter and VCA. The Delta and Sigma have trigger inputs for the envelope generators, while the Lambda cannot accept trigger inputs, but does give a trigger output.

The Pro-1 monophonic synthesizer, from Sequential Circuits, has a jack which accepts an input pulse for the arpeggiator or sequencer (this overrides the timing pulses coming from the LFO which normally drive the arpeggiator or sequencer). This is a direct sync type of jack, where one pulse equals one beat. (Also note that the Pro-1's envelope detector can generate trigger pulses from an externally applied audio input.) The Sequential Circuits Polysequencer, when run in the single-step mode, also accepts a direct sync pulse for synchronization effects.

Of course, other manufacturers make equipment which may be synchronized. PAIA's Proteus lead synthesizer has a computer port which may be accessed synchro-sonically for automatic retriggering and sequencing, while their Hyperflange Plus Chorus delay unit (which I designed) includes a sync pulse input for synching the LFO to the tempo of a given composition. Also, the voltage-controlled ADSR I explained how to build in the May '82 issue of *Keyboard* has an input which accepts retriggering pulses.

Interesting enough, even signal processors are getting into the act. Lexicon's new PCM-42 delay unit lets you program a pulse output which is a multiple of the clock frequency, thereby giving pulses which are directly related to delay time. Why is this useful? Well, suppose you want the PCM-42 to deliver a two-second delay. Unfortunately, it's quite hard to get a feeling of tempo from a rhythmic period which is this long; however, you can program a pulse from the PCM-42 which is a multiple of this rather long time period. You could easily program these pulses to appear every two seconds, every second, every half-second, every quarter-second, and so on. (There's even provision for programming some non-standard time intervals as well.) These pulses can then either serve as a metronome or drive devices such as rhythm units. Thus, the echo time can serve as a time base in a composition where you want rhythm and echo time synchronized.

Technical Considerations. Unfortunately, not all of these systems use compatible triggers. The Korg keyboards require a switch closure to ground for triggering — in other words, the input normally sees a high impedance, which must go to ground to give a trigger pulse (Moog equipment generally works in the same way). The Oberheim system works similarly to the Korg system; however, it has quite a low impedance (about 1k) input and therefore cannot be driven directly from something like a CMOS logic output.

The Pro-1 synthesizer requires a positive-going pulse, but that pulse must be around 10 milliseconds wide since the internal computer is often tied up doing other things, and 10ms seems about right to guarantee that the computer catches a pulse. The Roland TR-606 and TB-303 require a positive-going pulse, but in this instance the pulse can be very thin (1ms or even less) and the units will still "grab" this pulse. The PAIA Proteus synthesizer requires a circuit which resembles the Korg or Oberheim interface if you're going through the computer port. In Part 2, we'll discuss some interfaces which allow us to adapt the Master Synchronizer to a variety of triggering standards.

Getting A Click Track On Tape. So far, everything we have discussed is great for live performance. But what if you work in the studio, and want to synchronize overdubs?

You might think this is not a problem — after all, if a piece of equipment is chugging along at 120 beats per minute, then it will still be chugging along at the same speed when you're doing an overdub, right? Well, yes and no. While the tempo will be very close, even subtle variations caused by AC line voltage fluctuations or drift in the synthesizer or drum unit's clock generator will probably be sufficient to put you out of sync at some point during the composition. So what we need to do is create some kind of timing standard on one track of the tape recorder. During overdubs, if we feed this timing information into the sequencer, drum unit, or synthesizer we wish to overdub, everything stays locked in sync to the timing information stored on tape.

There are many ways to put timing info on tape. The system we'll be describing next month takes a "brute force" approach, which simply puts a pulse on tape at a multiple of the basic tempo. We then recover this signal, condition it, and put it into our synchro-sonic circuitry. This works fine if the timing track isn't too high with respect to frequency.

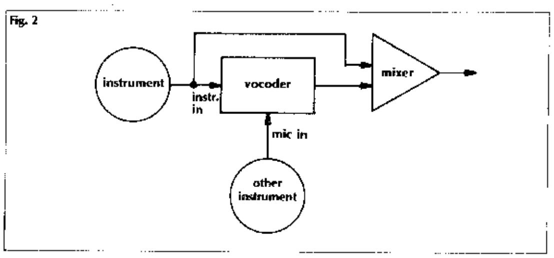
The Oberheim DMX drum unit and DSX sequencer use a more complex approach by dumping a 2.4kHz sine wave onto tape, which jumps up an octave to 4.8kHz to indicate each timing pulse. The reason for putting a frequency-shifted sine wave on tape is that Oberheim feels this produces more reliable results than the brute force method. These frequency-shift pulses occur at a rate between 40 and 400Hz. Since the DMX standardizes on 48 pulses per quarter note, this means that the DMX covers a total range from 25 beats per minute to 250 beats per minute.

One problem with systems which put a tone on tape is that should you use variable speed, the frequency of that tone changes. Fortunately, the Oberheim will accept tape speed changes up to ±20%, but unlike the system we'll be talking about next month, it will not work with double speed/half speed techniques.

Experiments To Get You Started. There are many sync effects you can create using common synthesizer modules and signal processors. To get you started, here are some examples which are intended for a home studio context, but will work live as well.

• *Vocoding.* Hook up a vocoder to your keyboard in the normal way, but instead of feeding in a microphone and talking into it, sample some rhythmic instrument such as rhythm guitar, bass, or drums. When hooked up in parallel to a direct signal (see Fig. 2) and mixed in subtly, the vocoder will cause the timbre of your keyboard sound to vary in relation to the other instrument.

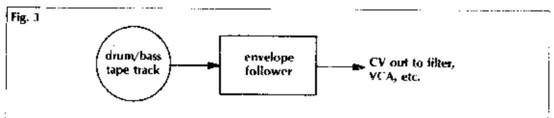
Fig. 2



Incidentally, if you haven't heard these kinds of effects before, you might presume that processing one instrument with another would produce a gimmicky sound. As it happens, if the modifying action is set for subtle changes, the synchro-sonic effect becomes almost subliminal. I think the reason for this is that if two or more events happen at *exactly* the same time, the ear can't really differentiate between the two, and instead hears an overall texture rather than a single "gimmicky" sound.

• *Envelope follower following other instruments.* Try enveloping-following drum and bass tracks (see Fig. 3), and use this control voltage to *subtly* alter the center frequency or resonance of your synthesizer's filter (inverted envelopes are also extremely useful). As the drummer plays or the bass player plucks, you will have an associated filter change which locks you into the groove set by the other player.

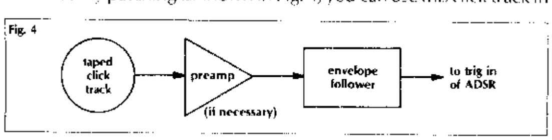
Fig. 3



• *Envelope-following special purpose tracks.* There's no reason why you can't envelope-follow a track put temporarily on tape strictly for the purpose of being envelope-followed but which gets erased before the final mix. For example, on one cut I recorded a drum track first. I then added several overdubs, with each overdub having some parameter (filter, VCA, modulation, etc.) controlled by envelope-following the drum track. The last track was a melody line which was not synchronized, and was recorded over the drum track. Despite the fact that the drum track was erased before the final mix, the piece had an exceptionally rhythmic feel.

• *Envelope-following the click track.* Many musicians will put down a click track using an acoustic or electronic metronome as a timing reference. By patching as shown in Fig. 4, you can use this click track in

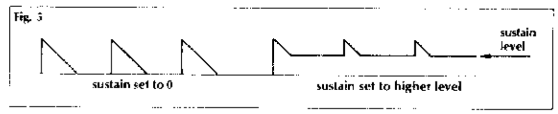
Fig. 4



conjunction with an envelope generator such as the VC-ADSR presented in the May '82 issue of *Keyboard* (note additional information on this project in the July '82 Letters column) to create a decay gate. Simply turn the gate delay, attack, sustain, and release controls to 0, while holding the gate high by patching J5 (the gate input) to a source of +5 to +15V DC. Then patch the envelope follower output to J7 (the trigger input) of the ADSR (you may need to amplify the click track if it lacks enough level to trigger the ADSR). The ADSR "A" and "D" cycles will trigger with each pulse; adjust the Initial Decay control for the desired amount of decay. If you want to get fancy, envelope-follow another intrument, and use that voltage to control the ADSR's initial decay control voltage input.

By the way, if you turn up the sustain control, you'll have a constant "reference" voltage coming out of the ADSR, with each trigger adding decaying peaks (see Fig. 5).

Fig. 5



• *Double-time click track.* You say your metronome click track is quarter notes, and you want eighth-note modulations? No problem. Feed the click track into a delay line set for equal amounts of straight and delayed sound (Fig. 6a) with no regeneration, and then patch into the envelope follower (again, you may need an amplifier after the delay line to bring the click up to the suitable level). Adjust the amount of delay for exactly one-half the period between clicks. If you get it set

Fig. 6A

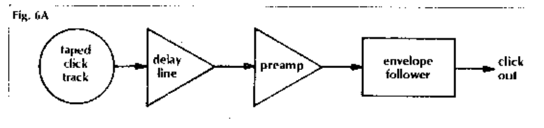
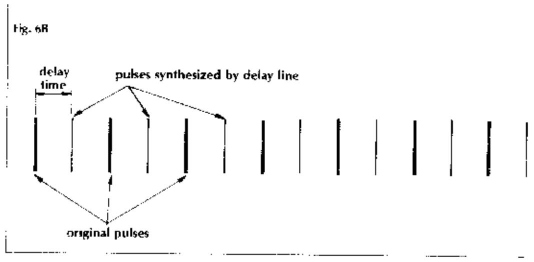


Fig. 6B

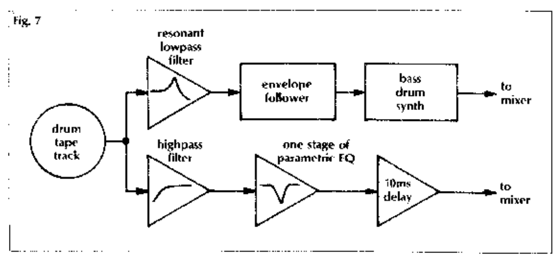


just right, you'll synthesize the extra click you'll need (Fig. 6b).

• *Bass drum recovery system.* This is not something you'd need to do every day, but it does demonstrate an unusual sync application.

I was working on a song using a drum machine which sounded fine — except that the bass drum sound was weak and just didn't "cut." So I whipped up a better bass drum sound on a synthesizer, and rigged up the patch shown in Fig. 7. A highly resonant lowpass filter was tuned

Fig. 7



right in on the bass drum frequency, thus filtering out the rest of the drum sounds (I used the Multiple Identity Filter that I explained how to build in my Electronic Projects column in the Nov. '79, Dec. '79, and Jan. '80 issues of *Keyboard*, but you could also use a sharp bandpass filter). I followed the envelope follower with a pulse conditioner, both of which are part of PAIA's "The Drum" (slightly modified for this application so that it could drive my bass drum module). The pulse conditioner cleans up the rough output of the envelope follower and turns it into a sharp trigger signal suitable for triggering a synthesizer. The output of the bass drum synthesizer went into one channel of the mixing board.

The next step was to feed the rest of the drum track through a highpass filter to remove the majority of the original bass drum sound, and then through some parametric equalization (I used one stage of the parametric I explained how to build in my article in the Apr. '81 issue of *Keyboard* — now being offered by PAIA as their 6760 equal-

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izer) to get rid of any remaining bass drum component. I fed this into the console, and surprise — the new bass drum hit just a little later than the rest of the drums, which was very distracting. After a little head-scratching, I realized that the filter and envelope follower were adding about 10 milliseconds of response time. So I patched in a good digital delay and delayed the original drum track (minus the bass drum) by 10ms so that it synched up with the new bass drum. The result? A drum track with the original weak bass drum surgically removed and seamlessly replaced with the stronger-sounding kick.

• *Using the Master Synchronizer module to create rhythmically timed control voltages.* First, you take the Master Synchronizer module . . . oops, I'm getting ahead of myself. We're still in Part 1 of this article! Well, I guess we'll just have to wait until next month before getting to this particular application. See you then.

NOTES

Craig Anderton's Electronic Projects, No. 8

SYNCHRONIZATION, Part II

Build A Master Synchronizer

LAST MONTH, WE discussed the basics of synthesis and synchronization, along with some practical examples. This month, we'll cover how to build the Master Synchronizer, plus some inexpensive and practical accessories to go along with this module.

I do feel I should give a warning, though. This is not too complex a project to build, but learning to use it is another matter. You'll need to devote a certain amount of effort, trial-and-error, and experimentation to get it working properly, let alone working at its maximum potential (luckily, though, once you become familiar with the device it is quite trouble-free). While I hesitate to say this is a project for advanced experimenters only — I don't want to turn off the patient beginners — it must be emphasized that unless you've worked with click track retrieval systems before, you're entering a whole new world. If you're patient, logical, and careful, the Master Synchronizer will treat you very nicely. Just don't expect to master it in a few minutes. You may need several days of experimentation before you figure out how to make best use of this module in your particular setup.

What It Does. In the studio, the Master Synchronizer module dumps timing information on tape and retrieves it (like a glorified click track). Once retrieved, this timing information is separated out into a number of rhythmically useful outputs, which may then drive devices with sync inputs (drum machines, sequencers, envelope generators, etc.) when doing overdubs, thus letting you sync these devices together. You may also drive some novel synchro-sonic control voltage generators, described later, which are inexpensive and very useful musically. There are a number of convenience features as well, such as simultaneous square wave and pulse outputs, choice of +5V or +10V outputs, metronome, and — well, there's so much stuff that after some introductory material about logic circuits, we'll just start describing the circuit. By the end of the article you should have a good idea of what the Master Synchronizer can do for you.

Craig Anderton is a highly respected expert in musical electronics, the author of two books, Electronic Projects For Musicians *and* Home Recording For Musicians, *the editor of* Polyphony *magazine, a longtime columnist for* Guitar Player, *and a former columnist for* Keyboard *as well. His most recent articles for* Keyboard *were "Delay Lines" (Aug. '82), "Signal Processing Chains" (Oct. '82), and the first half of his project on synchronizing keyboards, sequencers, and drum machines (Jan. '83).*

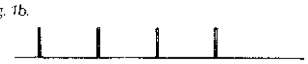
Logic Basics. This particular module uses digital circuitry. Digital logic IC outputs only have two states: on (output equal to supply voltage, also called the high state), and off (output at ground potential, also called the low state). The IC inputs recognize only high or low signals as well.

Most music devices (sequencers, arpeggiators, etc.) require a low-to-high transition in order to be triggered; others require a high-to-low transition. So, suppose we have digital circuits generating a square wave, as shown in Figure 1A. If this feeds the input of a sequencer which requires a low-to-high transition to advance one step, the sequencer will advance at each *rising edge* of the square wave. If these rising edges occur at a rhythmically interesting interval — such as every quarter note — the sequencer will advance with each quarter note.

Fig. 1a.

Unfortunately, feeding this waveform into another sequencer which requires a high-to-low transition will result in the two sequencers being out of sync, since one will trigger on the rising edge of the square wave, and the other on the falling edge. We can get around this problem by generating a narrow pulse wave instead of a square wave (see Fig. 1B). Now the rising and falling edges occur so close together, they are for all practical purposes simultaneous. Thus, we can have events occur at virtually the same time, whether a device triggers off of the rising edge or the falling edge.

Fig. 1b.

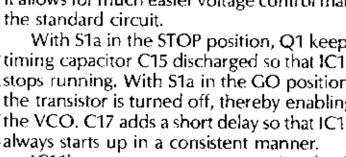


The Master Synchronizer generates pulse and square waves simultaneously at the most useful rhythmic intervals — sixteenth-notes, eighth-notes, quarter notes, half notes, whole notes, and some triplets as well. There's also a thirty-second note output. These are all synchronized to a master clock, which may be either generated internally by the Master Synchronizer, or stored on tape and recovered by the Master Synchronizer.

How It Works: Clock Section. The following is a bit technical, but don't let that scare you off. I think you'll find that the first time through, the whole concept will make little (if any) sense unless you are well versed in electronics. However, after you've reached the end and realized how the Master Synchronizer is supposed to be applied, the material presented in the beginning should fall into place when you re-read it.

Referring to the block diagram (Fig. 2), there is a master clock. When recording, the clock can send pulses to a tape recorder.

Fig. 2.



These pulses may then be recovered by the tempo recovery circuitry. A logic section processes this stream of pulses to create a variety of musically useful rhythmic outputs. This logic section may either receive pulses from the tempo recovery circuitry (switch in EXT position), or, for live use, receive pulses directly from the clock (switch in INT position).

The heart of Figure 3 is IC11, the master clock, which covers the range of 20 to 200Hz. IC9D is a control voltage mixer which sets the frequency of IC11. R38 determines the initial tempo; this can be sped up by feeding a positive voltage into J2, or slowed down by feeding a negative voltage into J2. Note that IC11 is configured in a non-standard way, where ground is referenced to the negative supply. This technique was first described by John Simonton in *Polyphony* magazine, and it allows for much easier voltage control than the standard circuit.

With S1a in the STOP position, Q1 keeps timing capacitor C15 discharged so that IC11 stops running. With S1a in the GO position, the transistor is turned off, thereby enabling the VCO. C17 adds a short delay so that IC11 always starts up in a consistent manner.

IC11's square wave output at pin 3 feeds IC9B. This stage turns the square wave into a 2ms wide pulse. This pulse goes to two places: the CLOCK OUT pot, which feeds the pulses to a tape recorded via J3, and the INT/EXT clock switch (S5). With S5 in the INT position, the logic section receives its clock pulses directly from IC9B's output.

IC10 and IC9A form the tempo recovery circuitry, which retrieves the clock pulses put

Fig. 3.

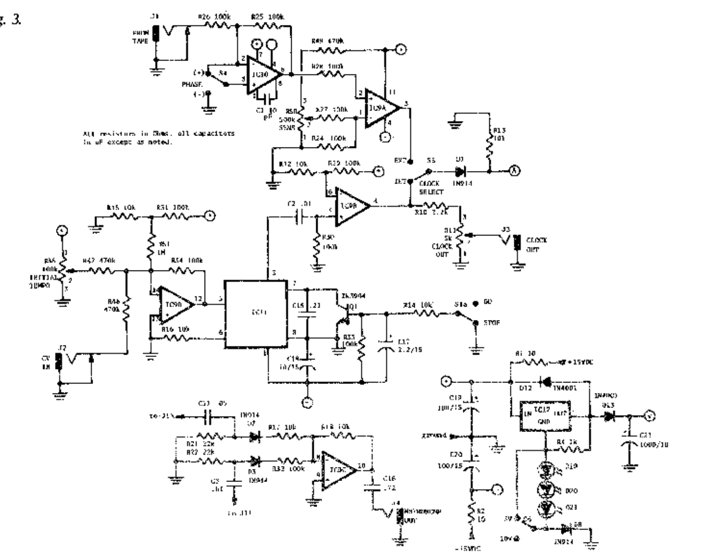
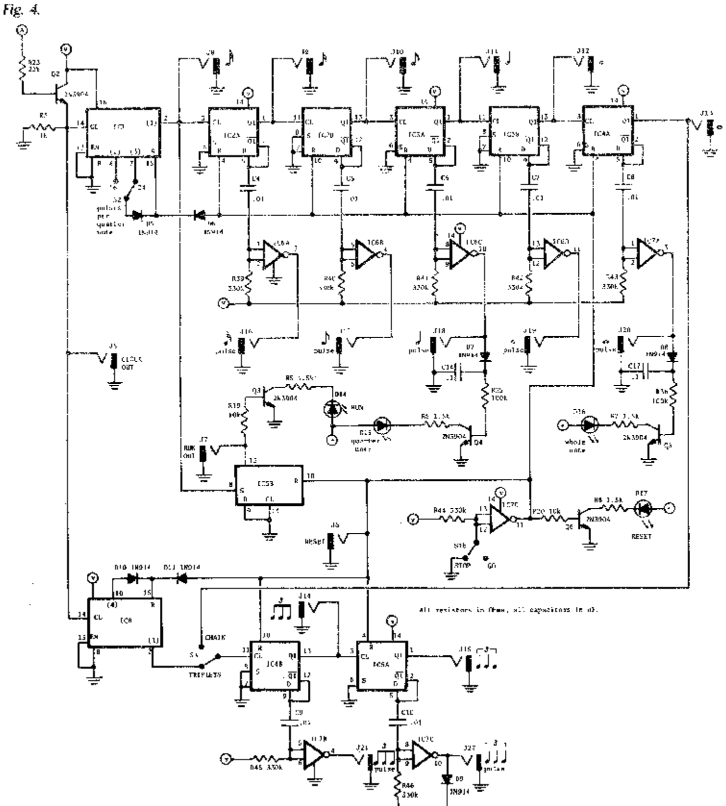


Fig. 4.



on tape. Since some tape recorders invert the phase of recorded signals and some do not, IC10 lets you match the Master Synchronizer to either type of signal (there's more on this later) via S4. IC9A then shapes up these clock pulses into a form which can drive the logic section. D1 prevents any negative voltages from either IC9A or IC9B's output from getting into the logic section, since the logic section only wants to accept pulses which alternate between ground and the positive supply.

The circuitry built around IC12 and IC9C will be discussed later.

How It Works: Logic Section. First, note that Point A on Figure 3 connects to Point A on Figure 4, the logic section.

As discussed last month, several commercially available units operate on a standard where 24 pulses occur between quarter notes. (Note that some units require 16 pulses per quarter note, and the Master Synchronizer provides for this. However, for simplicity we'll just look at the 24 pulses per quarter note operation for now). If 24 pulses equal one quarter note, then 12 pulses would occur between eighth-notes, 6 pulses between sixteenth-notes, and 3 pulses between thirty-second notes. Therefore, if we use a divider circuit to divide the incoming pulse stream by 24, the output of this divider would consist of pulses occurring every quarter note. But of course, there's more to life than quarter notes, so the logic section includes a number of dividers. IC1 in Fig. 4 divides the incoming pulse stream by 3, which means an output pulse occurs at its output (pin 2, or J8) for every three incoming clock pulses. As a result, this output consists of pulses whose rising edges occur at thirty-second-note intervals. IC1's output also feeds IC2A, which now divides this series of thirty-second-notes by two. Therefore, IC2A's output (J9) consists of square waves whose rising edges occur at sixteenth-note intervals; this output also feeds IC2B's input. IC2B again divides by two, producing eighth-note outputs. In a similar manner, IC3A produces square waves occurring every quarter note (incidentally, by now we've divided the incoming pulse stream by 24), IC3B produces half-note outputs, and IC4A produces whole-note outputs.

With S2 set to 16, IC1 divides by 2 instead of 3. This lets the Master Synchronizer accommodate 16-pulses-per-quarter-note devices, as well as the far more common 24-pulses-per-quarter-note devices.

Since we often want thin pulse outputs for triggering applications, consider IC6A. This stage takes the square wave output from IC2A and *differentiates* it, thereby creating a 2ms wide pulse at its output (J16). Similarly, IC6B, IC6C, IC6D, and IC7A create pulses from square waves produced at the outputs of IC2B, IC3A, IC3B, and IC4A, respectively.

What about IC8? This divides the clock frequency by four, and in conjunction with IC4B and IC5A, produces eighth-note triplet or quarter-note triplet square waves at J14 and J15 respectively (pulse versions of these appear at J21 and J22). However, in some applications you might not need triplets, but you might need intervals longer than whole notes. With S3 in the "triplet" position, J14 and J15 produce outputs as described above.

In the "chain" position, IC4B and IC5A feed off the output of IC4A, and produce outputs every two measures (J14/J21) and every four measures (J15/J22) respectively.

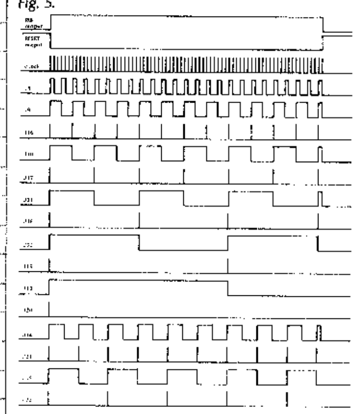
IC7D resets the entire logic section. Resetting restores all logic stage outputs to their low state, which means that when you start up the system, all pulses will occur with the correct synchronization (providing the system has been reset first). Resetting occurs automatically when the STOP/GO position is switched to STOP.

IC5B produces a logic high signal at J7 when the system is receiving clock pulses. Some devices need this run signal before they will accept external clock pulses.

There are also three monitor lights which help us understand what is happening with the system (thank heaven for small favors). D15 flashes every quarter note; D16 flashes every whole note; and D18 flashes every quarter-note triplet with S3 in the triplet position, and once every four measures with S3 in the chain position. D17 indicates that the system is in the reset mode, while D14 indicates that the system is running (i.e., that it is receiving clock pulses).

As a reference which should make the above clearer, Figure 5 shows the timing relationships between all outputs.

Fig. 5.



Complication: Some systems are designed to accept logic signals which alternate between ground and +5V, while others accept logic signals which alternate between ground and +10V. Our solution is to bring all power supply terminals for the ICs in the logic section to a separate power line buss, called "v." This is fed from regulator IC12 (see Fig. 3). IC12 takes the +15V supply voltage and, with S6 in the 5V position, produces +5V across capacitor C21 for feeding the logic section. With S6 in the 10V position, IC12 produces +10V across C21. Note that D19-D21 light up with S6 set to 10V; however, that is only coincidence, since we are not using these LEDs for illumination, but rather as voltage references to draw a specific amount of current through IC12's ground terminal. D12 and D13 are protection diodes, which let you do dumb things (like short the power supply to ground) without causing damage. D8 raises the output voltage a bit to compensate for the voltage drop occurring through D13.

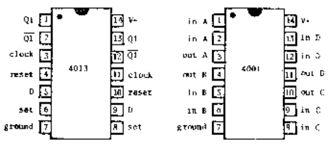
All that's left to discuss now is the metronome circuitry based on IC9C (see Fig. 3). This samples the signals from J11 and J13, and produces a loud click on the downbeat and a softer click every quarter note. You may make the quarter note click louder by lowering the value of R32.

Whew! So much for theory. Here are some notes on building the Master Synchronizer.

Construction Tips. This is not an easy project, but it's not too critical either. For those who don't wish to build from scratch, a parts kit is available (see parts list). Here are the major things to remember:

- Use sockets for all ICs. For best results, use a printed circuit board.
- Do not plug in any of the CMOS ICs in the logic section until the unit is completely wired. Handle these ICs with care, as they can be destroyed by static electricity discharge. Keep them in their protective foam until needed, and touch the circuit board ground line before removing the ICs from their foam and inserting them in their sockets.
- Use a well-regulated, ±15 volt power supply capable of delivering at least 50mA. Connect the +15V line to the point near IC12 marked "+15VDC," the −15V line to the point near IC12 marked "−15VDC," and the ground line to the point near IC12 marked "ground." All points marked with a circled "+" connect to each other. All points marked with a circled "–" also connect to each other. All points marked with a circled "v" connect together, and so do the two points marked "A."
- The pot terminals are numbered as follows: Looking at the back of the pot (*i.e.*, with the shaft pointing away from you and the terminals pointing down), the left hand terminal is 3, the middle terminal 2, and the right hand terminal is 1. R38 is an audio taper pot hooked up in reverse, so that turning clockwise produces a slower tempo. Granted, this is non-standard, but the resulting taper makes it much easier to set tempos.
- Be careful with your grounding. If possible, run a single ground wire from the Master Synchronizer circuit board to the metal front panel (or enclosure), which should connect to a large metal rack frame which connects to the power supply ground. If the rack frame is not metal, connect a wire from the front panel or enclosure to the power supply ground. These ground wires should be fairly heavy gauge.
- Figure 6 shows a suggested front panel layout for a 19" × 1.75" rack panel. Note that the six switches towards the right hand side are templated for slide switches; if you use toggle switches, then only drill the holes directly about the designations S1-S6. I would also suggest using small LEDs for maximum brightness and ease of construction.
- The pin numbers of the 4013s and 4001s are shown as wired in my prototype. However, a flip-flop is a flip-flop and a gate is a gate, so if it's more convenient, you may use any flip-flop for any other flip-flop. Figure 7 shows the complete pinout for these two chips.
- You're going to be running a lot of wires between the Master Synchronizer circuit board and the front panel. Be patient.

Fig. 7.



The results are worth it.

- Pay attention to the polarity of all electrolytic capacitors, LEDs, diodes, ICs, etc.
- If you're not familiar with soldering techniques, or have never constructed an electronic device before, this is not the place to start! Try tackling something simpler (such as one of the projects in my book *Electronic Projects For Musicians*) before attempting the Master Synchronizer, or purchase the parts kit and follow instructions *very* carefully.

Testing. Assuming that the Master Synchronizer has been correctly built, you're ready to check it out. Plug the metronome output into an amp, set S2 to 24, S3 to TRIPLET, S1 to STOP, S5 to EXT, and S4 to (-). S6 doesn't matter for now. The RESET light should be on. If not, re-check your wiring.

Now switch S1 to GO. The RESET light should go out, but the RUN LED will not come on since S5 is set to EXT, and at the moment, there is no external clock signal. Next, switch S5 to INT. After a very brief pause, the RUN LED will light up and you should hear an output from the metronome.

Vary the tempo control for the desired tempo. LEDs D15, D16, and D18 should blink with the tempo. If all is in order, the Master Synchronizer is now ready for live use. Now let's move on to tape interfacing techniques.

Tape Interfacing. It will take you a bit of effort to set levels properly, so don't get discouraged if you encounter problems at first. This is a fairly critical aspect of the Master Synchronizer's operation, but once you've established optimum operating levels, the thing is pretty much trouble-free.

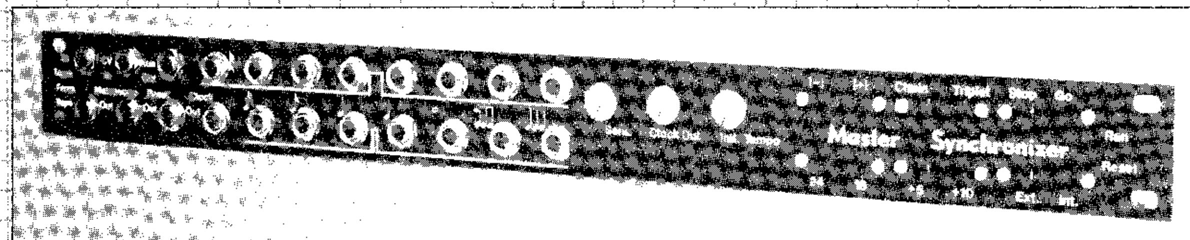
Here's the sequence of events for loading a click track on tape. *It assumes that your tape deck has three heads, and that you may monitor off both the playback head and the record head (sync mode).* Recorders which do not have three heads will require some slightly different procedures, as described later.

1. Patch J3 to a tape track input, and J1 to the same tape track output. The track output should be set for tape (*not* source). The recorder's input and output level controls should be initially adjusted so that a 0 VU signal in produces a 0 VU signal out. Listen to the metronome output through an amp.

2. Set S1 to STOP, S5 to EXT, the SENS control fully clockwise, and the CLOCK OUT control fully counter-clockwise. Load some tape, run the recorder, and press record.

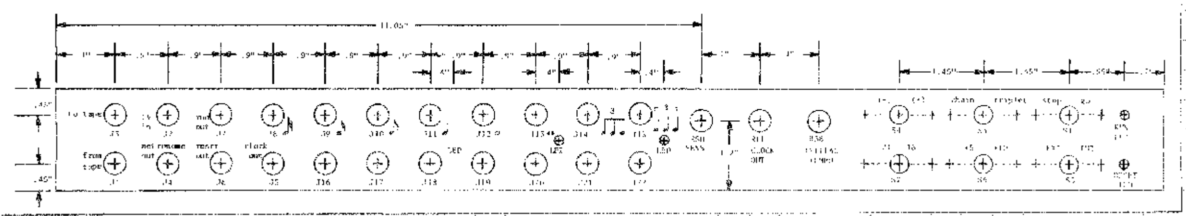
3. Set S1 to GO. The metronome will still be silent. Now turn up R11 until the VU meter reads about −10 to −7. The metronome may or may not be clicking by this point.

4. Adjust the tape channel's output control so that the metronome just starts clicking with the SENS control up full. In other words, if after step 3 the metronome is not clicking, turn up the output control. If after step 3 the



Master Synchronizer front panel.

Fig. 6.



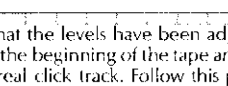
metronome is clicking, then turn down the tape channel output control until the metronome just stops clicking, then advance until the metronome just starts clicking again.

5. Switch S4 from (–) to (+). The clicking should go away. If not, leave S4 at (+) and repeat the previous procedures. The rule for setting this phase switch is *use the position which is most sensitive* (i.e., the position which requires the least amount of click track signal to kick the metronome into operation).

6. Turn the SENS control down to about 1/3 of its rotation.

7. In theory, you have now adjusted the levels for optimum operation. The important point to remember is that you want to keep the clock signal going to the tape at a constant -10 to -7 VU, and make any adjustments to match the Master Synchronizer sensitivity with the tape channel output control. If your deck does not have an output control, add a pot as shown in Figure 8 between the channel output and J1.

Fig. 8.



Now that the levels have been adjusted, rewind to the beginning of the tape and let's put on a real click track. Follow this procedure whenever you want to lay down a track

1. Set S1 to STOP and S5 to EXT. The RESET light should be on. Patch the metronome output to one of the tape tracks, and put this track in the record mode. Remember to turn off the record button for the click track channel!

2. Rewind to the point just before the click track begins. Set S1 to GO. The RESET light will go out.

3. Start the recorder. When the click track starts, the RUN light will go on. Check that the metronome signal is going on to the tape.

4. When the click track is finished, rewind back to the beginning of the tape. Put the track on which the metronome is recorded into the SYNC mode, and monitor it through your mixer. Send the metronome output (J4) from the master synchronizer to another channel for monitoring.

5. Set S1 to STOP and S5 to EXT. The RESET light should be on.

6. Rewind to the point just before the click track begins. Set S1 to GO. The RESET light will go out. When the click track starts, the RUN light will light up.

7. Now listen carefully to the metronome output from the Synchronizer, and the metronome output recorded on tape. They should be occurring in precise synchronization. If so, congratulations! Listen through to the end of the click track and make sure they remain in sync. Anything driven by the synchronizer will now be in sync with the click track.

using the Master Synchronizer:

1. Start off with S1 on STOP, and S5 on EXT. Make sure that J3 is patched to the proper tape track, that this track is ready to record, and that you're monitoring the click track off the playback head.

2. Run the tape, press record, and wait about 10 seconds. Then set S1 to GO. The RESET LED will go out, and shortly thereafter, the RUN output will light up, indicating that the pulses are being successfully sent to the tape and recovered by the tempo recovery circuit.

3. Let the click track record for as long as necessary. When finished, turn S1 to STOP.

To verify the click track, do the following:

In Case Of Trouble. As I said, this is a pretty critical aspect of operation, so you may run into problems. Here are some possibilities:

- *The two metronome signals start off in sync, but then go out of sync at some point.* This could be due to either physical damage of the tape itself, from a transient (refrigerator turning on, etc.) getting into your system and creating an extra pulse, or from a dirty tape head. Try synching again. If the same problem occurs consistently, re-record the sync track. Always verify the click track before proceeding with a recording, and handle the tape gently to prevent physical damage. A drop-out on a vocal may not be apparent, but a drop-out on the sync track will be apparent to the Synchronizer.

- *Unit fails to sync, or acts erratically.* This may require some playing with the SENS control and tape channel output control. So that you better understand what's happening, consider Figure 9. This shows what the click track looks like on a scope coming off of two different tape decks; Figure 9A is a cheap cassette deck, Figure 9B an expensive 8-track. The dotted lines show the best spot for picking up the click track. The SENS control varies the height of this line. If set too high, should one pulse be at a slightly lower level than the others, then you will lose that pulse. If set too low, then you might pick up some garbage such as double pulses or noise. If you have consistent sync troubles, look at pin 2 of IC9A with an oscilloscope to see what the pulse looks like coming off the tape. Then look at pin 1 to see the reference level set by the SENS control. Adjust the SENS control so that it lies about where the dotted line does in Figures 9A and 9B.

Fig. 9a.

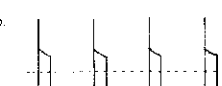
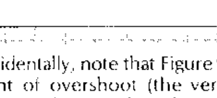


Fig. 9b.



Incidentally, note that Figure 9B has a fair amount of overshoot (the very thin line which extends upward). Make sure you don't try to use this as your click reference, since the overshoot pulse is so thin it may easily disappear if the tape has even a slight drop-out.

- Unit works reliably and steadily, but is consistently out of sync by a constant amount. In this case, make sure that you're monitoring the click track, metronome track, and any instruments you've recorded in the proper relationship to each other. Here's the way I do it:

1. I always monitor the click track off of the playback head. This is because when you're in the sync mode, leakage can occur between tracks. A strong leakage signal hitting the click track can cause real problems. The disadvantage of this approach is that you can only use synchronized devices during recording, not during mixdown when all instruments are being monitored from the playback head.

2. All subsequent tracks are laid down in step with the metronome. When listening to the metronome track, make sure it is in the sync mode. After laying down an instrumental part, put its track in sync (just like normal) when listening back. Of course, if your tape deck only allows you to monitor all the tracks from one head or the other, rather than letting you listen to one track from the record head and one from the playback head at the same time, you'll have to choose which method to use. If you monitor from the record head (sync mode), which is the only option you'll have on a two-head machine, you may have to increase the level of the click track somewhat so that it is quite a bit stronger than any leakage signals. If you monitor from the playback head, the second and subsequent tracks of music that you record will appear temporarily to be out of sync with previously recorded tracks, which can be a real brain-twister if you're trying to play anything live at the same time. But on playback, the problem will disappear.

3. I find it very helpful to record the metronome track on tape, and simultaneously monitor the metronome signal coming from the Master Synchronizer. This way, I can tell if any sync problems are occurring. 99% of the time, if the two do go out of sync, rewinding and trying again (without altering any of the controls) will solve the problem.

There are many additional points which could be made, but space is short. For example, after you become proficient at using the device, you can record the click track, a metronome track, and an instrument track simultaneously on the first pass. When monitoring off the playback head during mixdown, to avoid the out-of-sync problems mentioned above you can use a delay line and thus use the synchronizer during mixdown operations.

The Master Synchronizer invites experimentation, so experiment! It is a very flexible device, which is wonderful once you know what you're doing -- but getting to know it can have its moments of frustration. Relax, and take it easy. After a bit of practice everything should work just fine.

PARTS LIST:

Resistors

All resistors in ohms. 1/4-watt, 10% or 5% tolerance.

R1, R2	10
R3, R4	1k
R5-R9	1.5k
R10	2.2k
R11	5k linear taper pot (clock out level)
R12-R20	10k
R21-R23	22k
R24-R37	100k
R38	100k audio taper pot (initial tempo)
R39-R46	330k
R47-R49	470k
R50	500k linear taper pot (sensitivity)
R51	1M

Capacitors

Electrolytic capacitors may have *higher* working voltages than the ones indicated.

C1	5 to 10 pF ceramic disc
C2-C10	0.01 μF (mylar)
C11	0.05 μF disk
C12-C14	0.1 μF disk
C15	0.22 μF (mylar)
C16	0.22 μF (disk or mylar)
C17	2.2 μF @ 15V electrolytic
C18	10 μF @ 15V electrolytic
C19-C20	100 μF minimum @ 15V electrolytic
C21	1000 μF @ 10V electrolytic

Semiconductors

D1-D11	1N914 or equivalent silicon diode
D12,D13	1N4001 or equivalent power diode
D14-D21	Red LED (D14 may be green)
IC1,IC8	CD4017 decade counter/divider
IC2-IC5	CD4013 dual D flip-flop
IC6,IC7	CD4001 quad 2 input NOR gate
IC9	RC4136, XR4136, μA4136 or equivalent quad op amp
IC10	LM301, LM748, or equivalent uncompensated op amp
IC11	NE566, LM566, or equivalent VCO
IC12	LM7805 +5V voltage regulator
Q1-Q7	2N3904 NPN transistor

Mechanical parts

J1,J2	1/4" closed circuit phone jack
J3-J22	1/4" open circuit phone jack
S1	DPDT toggle or slide switch
S2-S6	SPDT toggle or slide switch
Misc.	Sockets for all ICs, front panel, wire, solder, knobs, etc.

Parts Kit Available: The following are available from PAIA Electronics (Box 14359, Oklahoma City, OK 73114): Punched and printed front panel (#6770fp), $17.88 plus $2.00 postage and handling; drilled and printed circuit board (#6770pc), $16.25 plus $2.00 postage and handling; complete kit of all parts including front panel and circuit board (#6770K), $74.95 plus $3.00 postage and handling. Oklahoma residents add tax.

SPECS:

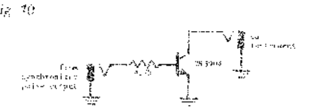
Clock range: 20-200 Hz
Beats per minute: 50-500 BPM (24 pulses/quarter note)
75-750 BPM (16 pulses/quarter note)
Logic output levels: switchable, +5 or +10V
Pulse output widths: 2 to 3ms
Recommended logic output load: greater than 2k ohms
Clock output level: 15V peak-to-peak
Current consumption: +35, -20 mA typical at ±15V

Interfacing With Commercial Equipment. As mentioned last month, not all devices follow the same triggering standards. While many devices (such as the Sequential Circuits Pro-1's arpeggiator) will trigger with either the square or pulse outputs from the Master Synchronizer, others require some kind of interface. Don't forget to select either +5 or +10V outputs, whichever is correct for your system.

The Master Synchronizer can also be used to drive many devices (such as Linn drum machines and various digital sequencers) which use divided sync. In the case of the Linn, which uses a 48-pulses-per-beat standard, you merely have to make a mental calculation to the effect that a quarter note on the Master Synchronizer will correspond to an eighth-note on the drum machine.

Figure 10 shows an interface for Korg, Moog, and Oberheim units. I built this particular interface right into a patch cord, and you might want to do the same. The same basic idea will also work for triggering PAIA's Proteus I envelope generators via the computer port. Connect the transistor's collector to pin 17 of the DB-25 connector feeding the port.

Fig. 10



The TR-606 Drumatix and TB-303 Bass Line from Roland have a five-pin connector on the back which accepts reset, run, and clock pulses; the Korg KPR-77 Programmable Rhythm has an identical five-pin arrangement. You would hook up the output from J5 to the clock input, J6 to the reset input, and J7 to the run output. For best results, connect a 10k resistor from pin 1 (run/stop) of the Roland unit's 5-pin connector to its positive supply. Also wire a momentary pushbutton from pin 1 to ground. Just before you start your synchronized run, push the pushbutton. This allows the 606 or 303 to start up in a consistent manner.

It is impossible to cover all possibilities for interfacing with commercially available equipment. Any shop which has the service manual for a given piece of equipment is the first place you should turn for advice on interfacing with the Master Synchronizer. Generally, though, you can feel free to experiment. It's doubtful that you will damage your equipment if you plug in, say, a square wave instead of a pulse.

Accessories. You can drive a bunch of devices with the Master Synchronizer outputs. I regularly drive a drum set, my synthesizer arpeggiator, Proteus envelope generators, and more. I'd like to go into this further, but this project is already longer than any of my previous articles. Perhaps if there is enough reader interest, we can run a sequel on application techniques, modifications, common questions, and so forth. In the meantime, here are some simple synchrosonic control voltage generators you can drive with the Master Synchronizer pulse outputs.

Figure 11 shows a vibrato signal generator. Feed a pulse into the input, and adjust the smoothing control for the smoothest vibrato effect. Note that this vibrato will be precisely synchronized with the tempo of the composition.

Figure 12 shows an inexpensive decay generator. Feed a pulse into the input, and adjust the decay control for the desired amount of decay.

Figure 13 shows an attack/decay generator. Feed a square wave into the input, and adjust the attack and decay controls to suit. Note that the attack time should not be set longer than the period of time that the square wave is high. Also, the decay control adds its decay during the time that the square wave is low. Thus, if it is set longer than the period of time that the square wave is low, the decay will never reach zero.

Figure 14 shows a four-step sequencer, driven by a pulse output from the Master Synchronizer. You can use this to create interesting stepped control voltages, which when processed through a lag processor become even more interesting.

There are many other possibilities — digital sine wave CV generators, pseudo-random CV generators, sample-and-holds (a great synchro-sonic effect, and the pulse is just perfect for providing the sample signal), and what this thing does for drum units is amazing. If only we had another 20 pages . . . oh, well.

The Master Synchronizer is probably one of the most significant projects I've ever presented. To the adventurous, it opens up a whole new world of synchronized effects and rhythmically complex music. Remember, be patient, take the time to learn how to use it, and the results should be as gratifying to you as they have been to me.

Fig. 11.

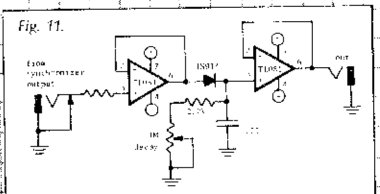


Fig. 12.

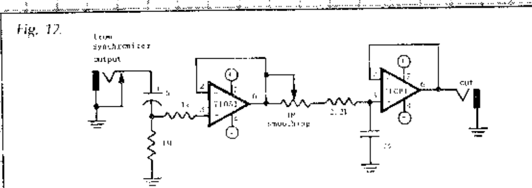


Fig. 13.

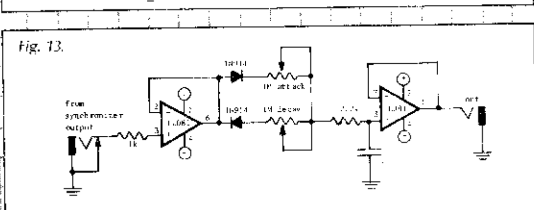
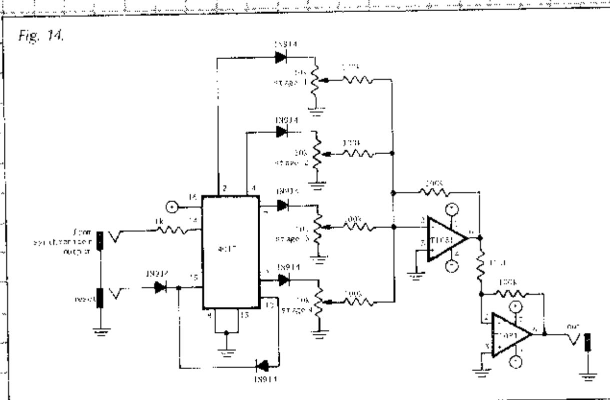


Fig. 14.



NOTES

!! IMPORTANT !!

Prior to beginning assembly of you new kit, check the supplied parts with the following parts list. BE DILIGENT.

QNTY	VALUE	DESC. (alternate markings)
		FIXED RESISTORS
2	10 Ohm	brown-black-black
2	1K	brown-black-red
5	1.5K	brown-green-red
1	2.2K	red-red-red
9	10K	brown-black-orange
3	22K	red-red-orange
14	100K	brown-black-yellow
8	330K	orange-orange-yellow
3	470K	yellow-violet-yellow
1	1 Megohm	brown-black-green
		CERAMIC DISK CAPACITORS
		(ALTERNATE MARKINGS)
1	10pf	10
1	.05 MFD	502
		POLYSTYRENE CAPACITORS
9	.01 MFD	.01K
3	.1 MFD	.1K, 104
2	.22 MFD	.22K, 224
		ELECTROLYTIC CAPACITORS
1	2.2 MFD/15V	Greater voltage ratings acceptable
1	10 MFD/15V	
2	100 MFD/15V	
1	1000 MFD/15V	
		SEMICONDUCTORS
11	1N4148 or 1N914	SILICON DIODE
2	1N4001-1N4004	POWER DIODE
5	TIL-209	RED LED
1	5.6V	ZENER DIODE
2	CD4017	CMOS DECADE COUNTER IC
4	CD4013	CMOS DUAL D FLIP-FLOP IC
2	CD4001	CMOS QUAD 2 INPUT NOR GATE IC
1	RC4136	QUAD OP-AMP IC

1	LM748	OP-AMP IC
1	NE566	VCO IC
1	LM7805	+5V VOLTAGE REGULATOR IC
7	2N4124	NPN TRANSISTOR

POTENTIOMETERS

1	5K	LINEAR TAPER POT
1	250K	AUDIO TAPER POT
1	500K	LINEAR TAPER POT

INPUT/OUTPUT JACKS

2	1/4"	CLOSED CIRCUIT PHONE JACK
20	1/4"	OPEN CIRCUIT PHONE JACK

SLIDE SWITCHES

1	DPDT	SLIDE SWITCH
5	SPDT	SLIDE SWITCH

MISCELLANEOUS PARTS

2	8 PIN	IC SOCKET
7	14 PIN	IC SOCKET
2	16 PIN	IC SOCKET
3	PUSH-ON	CONTROL KNOBS
12	4-40X1/4"	MACHINE SCREW
12	4-40	HEX NUT
3	3/8"-32	HEX POT MOUNTING NUT
1	3 PIN	PC MOUNT MOLEX POWER CONNECTOR
6 1/2 FT.		BARE WIRE
30 FT.	#22	STRANDED INSULATED WIRE
1	6770	RACK-MOUNT FRONT PANEL
1	6770	PRINTED CIRCUIT BOARD

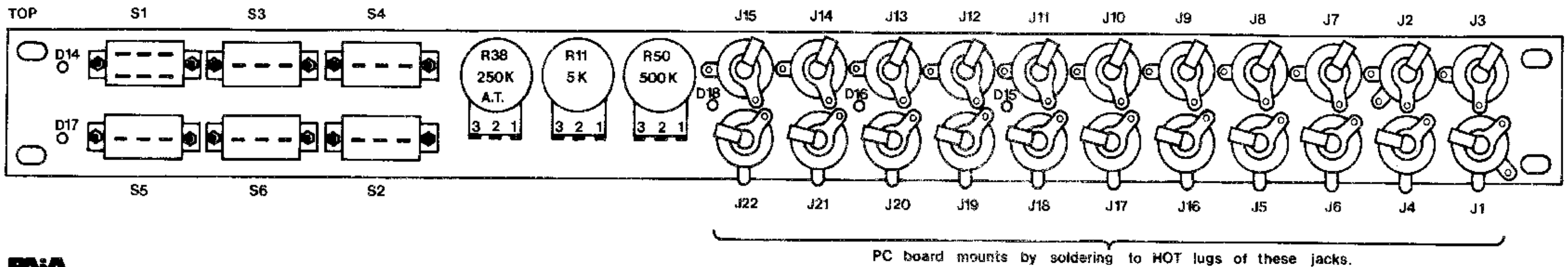


FIG. B1

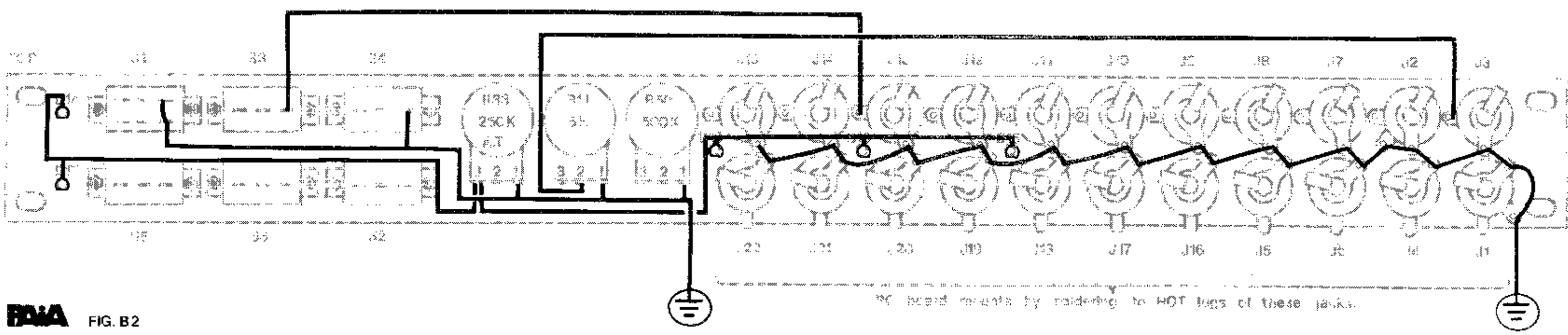


FIG. B2

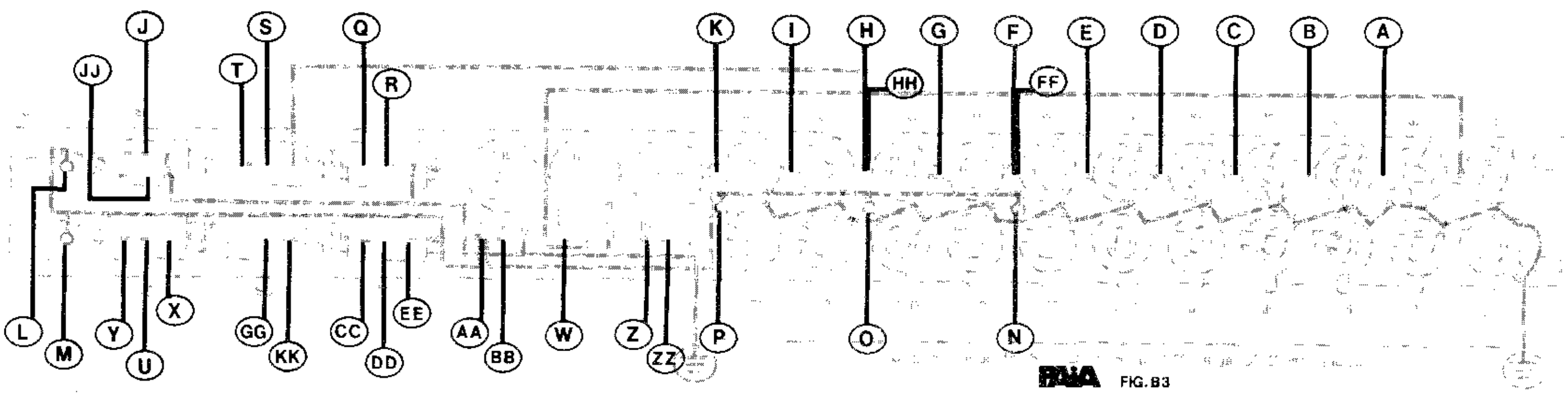
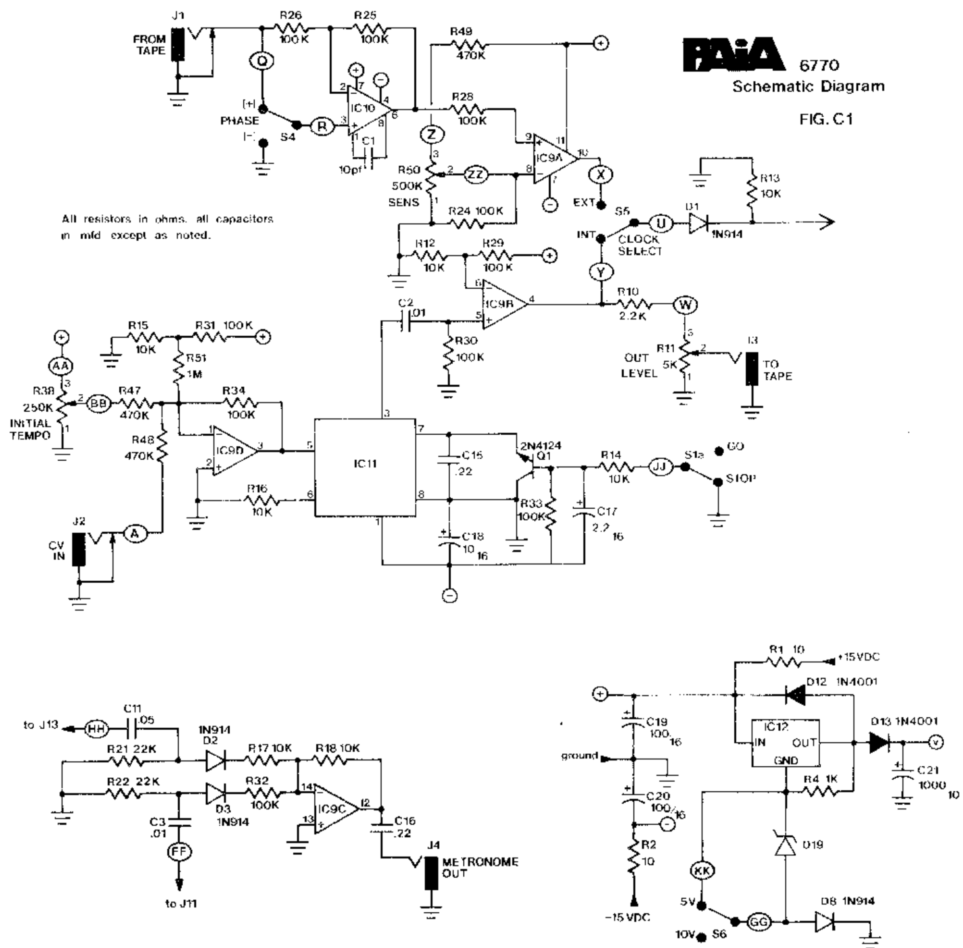
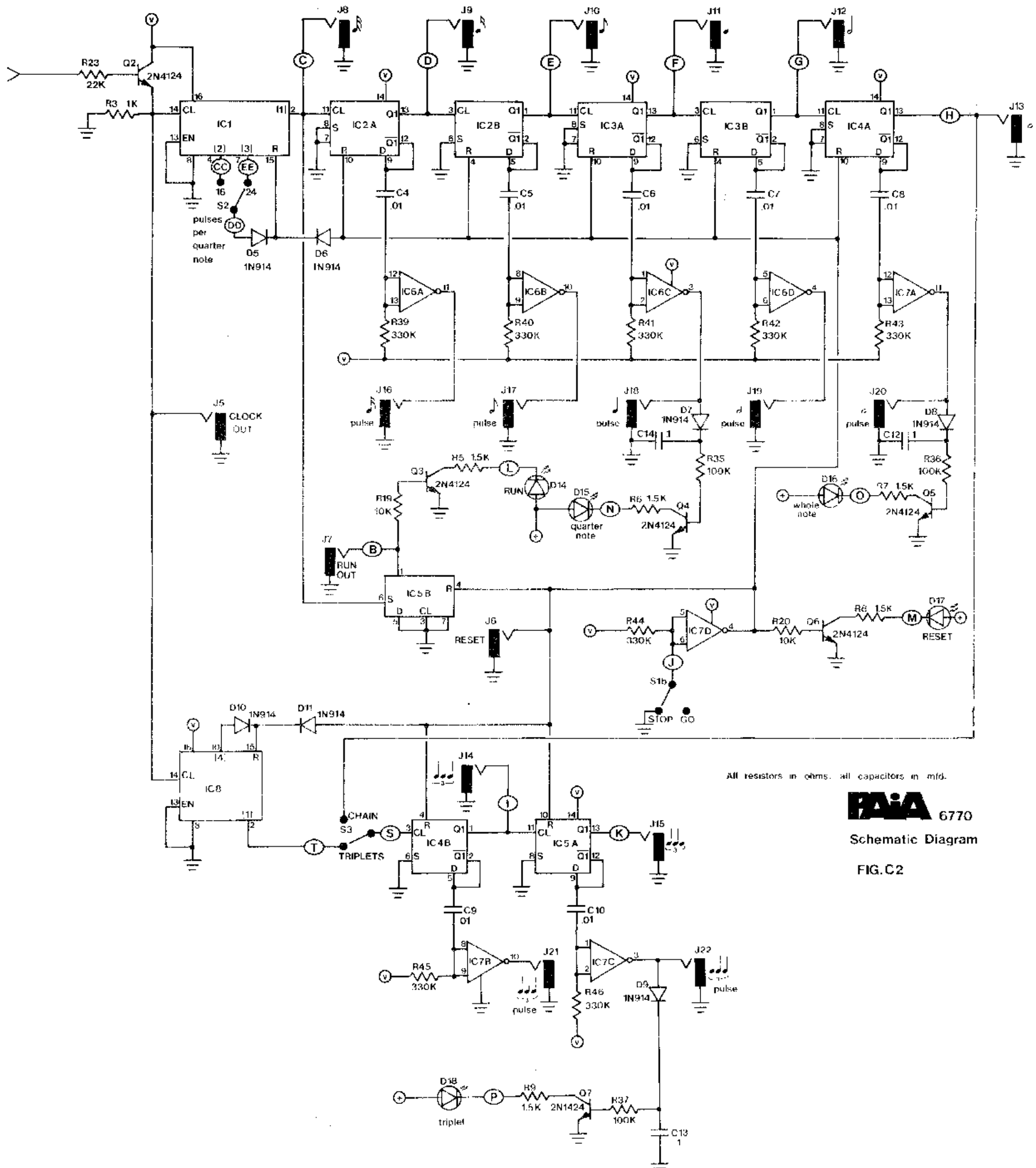


FIG. B3



PAiA 6770 Schematic Diagram

FIG. C1



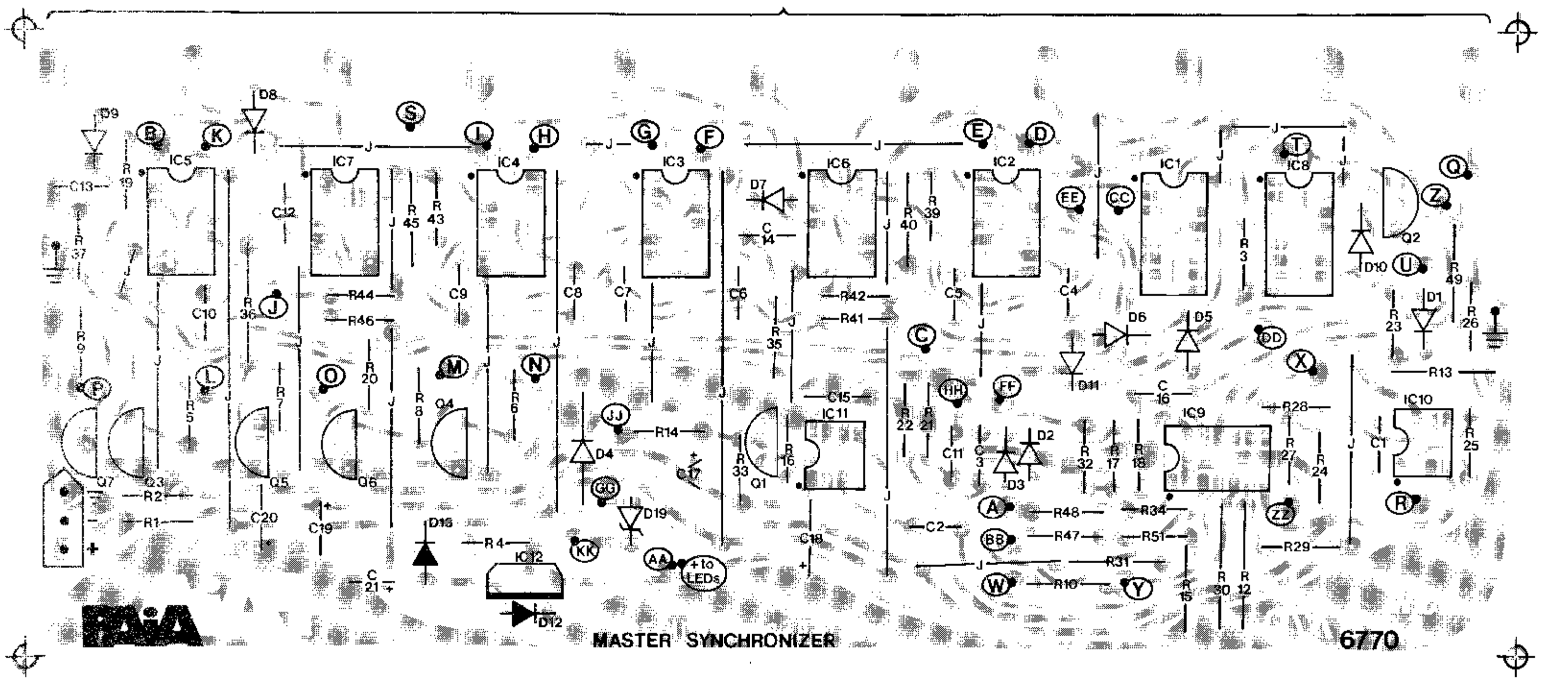
All resistors in ohms, all capacitors in mfd.

PAiA 6770

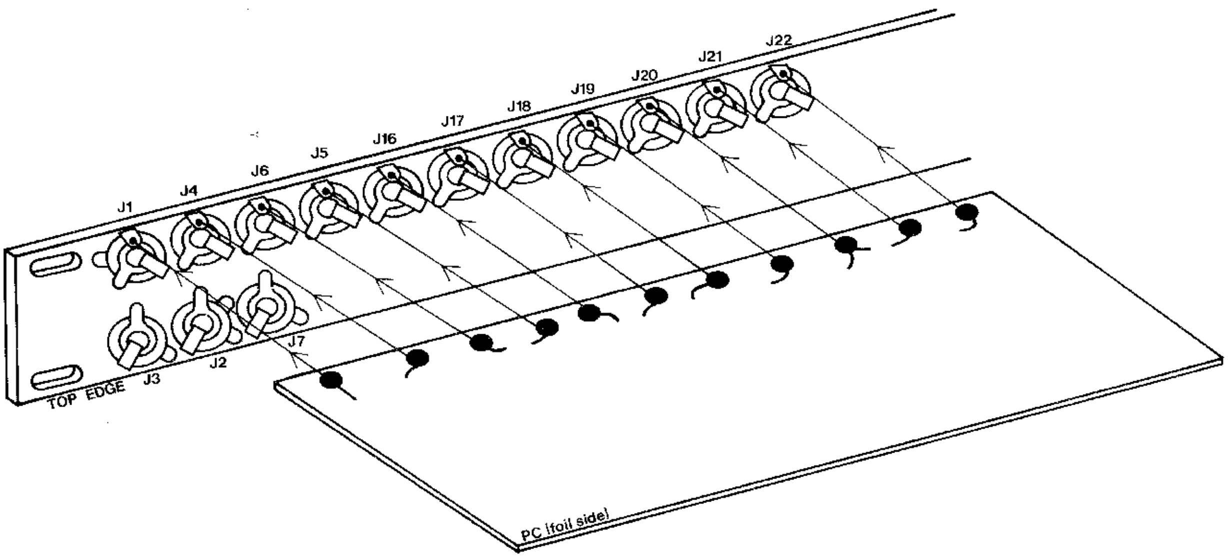
Schematic Diagram

FIG. C2

SOLDER THESE PADS TO JACK LUGS



Note: There are 23 bare wire jumpers [—J—] on the PC board.



The circuit board mounts on the BOTTOM row of jacks, as shown, by soldering the HOT lugs of the jacks to the respective solder pads on the foil side of the board. Be sure to tin the lugs and the solder pads first. It helps to leave the jacks untightened while mounting the board so that you can more easily align the lugs and pads. After the board is mounted, tighten the nuts on the jacks very carefully so that the jacks don't twist in their holes and tear the conducting foil from the board.