

*Making Music with the 566*  
By Thomas Henry

©2003, 2007 Thomas Henry

First Edition  
Second Printing (updated, prepared for PDF output)

Printed in the United States of America

A portion of Section 1.10 appeared previously in the article, "Build a Tunable Noise Generator," *Nuts & Volts Magazine*, November 1999, and is used here with the kind permission of its publisher, T & L Publications, Inc.

All rights reserved. Reproduction or use, without prior written permission, of any portion of this book is prohibited. The information contained herein is believed to be accurate at the time of publication. Neither the author nor the publisher assume any responsibility or liability for any inaccuracies or printing errors. The information contained herein is presented for illustrative purposes only. Neither the author nor the publisher assume any responsibility or liability for any consequences resulting from the use or misuse of any information contained within this book.

This and other Thomas Henry books are distributed by

**Magic Smoke Electronics**  
*www.magsmoke.com*

## About the Author

Thomas Henry is the author of over 100 articles and two books on the subjects of electronic music, microcomputers, astronomy, and caves. While in school he helped form the East Side Pharaohs (the Midwest's zaniest band) in which he played guitar and sand bass for nearly 18 years. After attaining in M.A. in mathematics, he taught at the collegiate level for ten years. His leisure time activities include bird watching, caving, amateur astronomy, magic, road trips to national parks, and flower gardening. The bat is his favorite mammal.

*Editors Note: In the winter of 2005, I was approached by Scott Stites. Scott is an avid electronics hobbyist who had been working with Thomas Henry testing and prototyping several circuits, some of which were slated to appear in a new book. Unfortunately, Thomas had recently decided to pull the plug on his business, "Midwest Analog." The decision was a good one for Mr. Henry, and it allowed him to pursue one of his other loves, teaching, full-time. Unfortunately, this left no distribution outlet for the Thomas Henry "Cookbook" series. These books had reached near-legendary status among synthesizer hobbyists, and many people wondered if they were going to join the ranks of the myriad other out-of-print books treasured by synth fans. Scott had followed my posts on the Synthesizer DIY mailing list during a particularly interesting discussion about copyright laws and whether copying was okay in cases where publications (like the Thomas Henry books) were no longer available. Despite all logic, Scott thought I might be a good choice to carry on distributing the books. I contacted John Mahoney, with whom I had worked before, and asked if he wanted to join in on a possible venture to publish Thomas' work. The reply was an immediate and enthusiastic "yes!" And so, Magic Smoke Electronics was born. Although we have plans to create additional books and circuit kits, if we do nothing other than provide an outlet for Thomas' existing body of work, we'll consider Magic Smoke a success. This book has been published with an absolute minimum of changes from the original. The section numbering has been changed slightly to make it easier to follow, and references to defunct web links have been removed or updated. Otherwise, these books remain true to the original works, and are a marvelous source of knowledge and inspiration. John and I would like to thank Mr. Henry for allowing us to continue publishing these books, and we hope you, the reader, will enjoy them as much as we have.*

Tim (Servo) Parkhurst  
March, 2007

# Contents

---

<b>Preface</b>	<b>v</b>
<b>Resources</b>	<b>vi</b>
<b>1 Design Collection</b>	
1.1 General Requirements.....	1
1.2 Wide Range Linear VCO.....	6
1.3 Exponential VCO.....	8
1.4 Triangle and Sine Wave Outputs.....	11
1.5 Pulse Width Modulated Output.....	14
1.6 Square Wave and Sub-Octave Outputs.....	15
1.7 Quadrature Square Wave Outputs.....	17
1.8 Ramp Wave Output.....	19
1.9 Adding Hard Sync.....	21
1.10 A Tunable Noise Source.....	23
<b>Bibliography</b>	<b>26</b>

# Preface

---

The 566 is an eight pin marvel, designed especially for voltage controlled oscillator (VCO) applications. However, over the years it's gotten just a bit of a bum rap. Many people will tell you that the 566 is hard to use, or that its range is too restricted or that the control voltage input is awkwardly (and backwards) referenced to the positive supply only.

Well, if all you ever do is go by the data sheet, then these comments would be true. But in fact, there are a number of slick tricks known only to a handful of designers that really unleash the power of the 566. Once you master these techniques you can coax the chip to flex its muscles, giving you:

- ♦ A 1:1000 sweep range (not the puny 1:10 span specified in the data sheet)
- ♦ Exponential response in addition to the standard linear response
- ♦ Direct voltage control, referenced to ground (i.e., an increase in voltage yields a higher frequency)
- ♦ Ramp, pulse and sine outputs, in addition to the usual triangle and square waves
- ♦ The ability to hard sync to another oscillator
- ♦ And more!

By the time you're done reading this guide, you should come away with a new respect for the 566!

Now I'm not going to claim that the chip is ideal for all musical applications. There's no doubt that dedicated integrated circuits like those produced by Curtis Electromusic Specialties (CEM Series) and Solid State Micro Technology (SSM Series) are still the quickest and easiest way to get a pro quality keyboard synthesizer up and running. But for other applications like electronic drums and control circuits, these special purpose ICs are simply overkill. Not only are they expensive, but they're often hard to find.

Enter the 566! While not as common nowadays, it remains the perfect choice for a low cost sound generator in lots of different musical situations – most notably synthesized percussion. I have used and continue to use it in all sorts of electronic drum circuits.

Like most of my books and articles, I wrote this initially to help me organize my own thoughts. Over the years I had stumbled across a number of interesting techniques for turbocharging the 566, some in magazine articles and some right at the workbench. I decided to pull together all my notebook scribbles and come up with a one-stop reference on using the 566 for my ideas and present them in an easy to use format. Like I say, I (rather selfishly) wrote this up to simplify my own work with the chip, but perhaps you'll find the results useful in your own projects.

I hope you'll like it.

Thomas Henry  
North Mankato, Minnesota  
June 2003

# Resources

---

The 566 VCO chip was originally released by Signetics back in the early 1970s. To give it its full name, it was known as the NE566. Later on, National Semiconductor second-sourced it as the LM566. As near as I can tell, the chip is no longer being produced, but it is still occasionally available from surplus and mail order electronic supply houses. Perhaps you even have some squirreled away in your own parts bin! It probably would have been (and would still be) a major selling item had the data sheets and application notes made it clear that you can wrench a great deal more power out of the chip with just a bit of tinkering.

Even though the manufacturer's data sheet (mentioned in the Preface) doesn't really make it clear just how useful the 566 can be, it still would be a good idea to procure a copy. If nothing else, you can check it for maximum ratings and so forth. And with the guide you now hold in your hand as an aid, you should be able to work your way past any of the deficits in the manufacturer's information.

*Editors note: The easiest way to lay your hands on the data sheets and application notes is by downloading them from the Internet. You can find a link to where these are stored on the WWW homepage of Magic Smoke Electronics.*

Magic Smoke Electronics: [www.magsmoke.com](http://www.magsmoke.com)

And don't forget to take a look at what other people have done with this chip. The Bibliography on page 26 will point you in the right direction. Electronic music circuit design has progressed quite a bit in the past three decades, so some of the earlier books and articles may show their age. Nonetheless, you'll frequently pick up a neat wrinkle of two by looking over another person's creation.

You'll note that the Bibliography cites several articles which appeared in various PAiA publications and some which were published in Nuts & Volts. If you'd like to add these valuable articles to your own design library, go to the following Web pages to check out back issue availability:

PAiA Electronics: [www.paia.com](http://www.paia.com)

Nuts & Volts Magazine: [www.nutsvolts.com](http://www.nutsvolts.com)

And oh, before I forget: many of the tricks in this guide work with the 8038 and other function generator chips, often with very little modification.

# 1.1 General Requirements

The 566 is actually pretty easy to use. The power supply options are flexible, and all of the building blocks within the chip are stable and predictable in their operation. This section is concerned with the general requirements you need to meet when designing with the 566. We'll start by painting the picture in fairly broad strokes, and only after that will we move on to the details.

The 566 implements a VCO in a typical way, with a Schmidt trigger / integrator combination. You're probably already familiar with the general idea from other music circuits that you might have encountered. A current is slowly dumped onto a timing capacitor (this is the integrator). When the voltage reaches a certain level, a Schmidt trigger comparator flips state, providing a discharge path. The voltage on the capacitor is gradually bled off and when the level falls sufficiently, the Schmidt trigger does its thing again and the cap begins to charge once more. A voltage controlled current source lets you modulate the output frequency under external control.

There are three major factors which determine the frequency of the oscillation: the timing capacitor just mentioned, a timing resistor, and a control voltage. As it comes from the factory, the control voltage option is linear and limited to a 1:10 sweep range. Fortunately, we'll see how to improve that dramatically in later sections.

Figure 1.1-1 shows the basic block diagram of the chip. If you're really interested, the data sheet for the 566 (from both Signetics and National Semiconductor) gives the complete schematic for what's inside. By my count, I make out 20 transistors, 10 diodes and 16 resistors. However, it isn't necessary to understand the schematic in order to use the 566. For now, let's just run around the pins to get familiar with what's there.

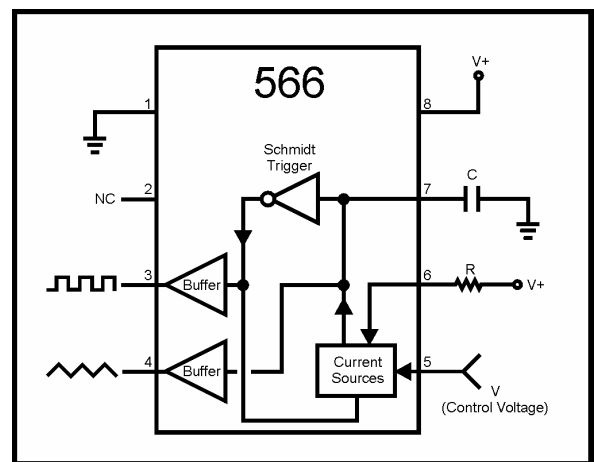


Figure 1.1-1  
Block Diagram of the 566

In the usual setup, pin 1 is grounded and pin 8 is connected to the positive supply voltage, which will be referred to as **V+** from now on. Later, we'll see how to cleverly use a negative supply voltage to obtain a marvelous improvement. The 566 can function properly on any supply voltage from +10V to +24V. By the way, the current drain is around 10mA or so in most circumstances.

Also note that it is possible to connect the IC across a dual supply (a bipolar  $\pm 5V$  for example) which has certain advantages with regard to referencing the square wave output to ground. This may be handy if your goal is to interface the oscillator to TTL circuitry. The manufacturer's data sheet gives the details if you're interested.

The timing capacitor mentioned earlier connects to pin 7. Its value will be referred to as **C** from now on in the formulas yet to come. A quick look at the block diagram confirms that you can tap the triangle wave off of this pin as long as you're careful to buffer it. But in most cases, it's simpler to just let the internal buffer do its thing and grab the triangle at pin 4.

The main timing resistor hooks up to pin 6 and will simply be called out as **R** in the formulas. According to the spec sheet, the value of R should lie between 2K and 20K. Note that a resistor tied to the supply is really just a trivial current source. So, it shouldn't surprise you that you can treat pin 6 as a current input. If it's desired to control the frequency of operation this way (eliminating R and injecting a current directly), then the data sheet dictates that the current must range between 100 $\mu$ A and 1000 $\mu$ A.

Incidentally, both C and R should be good quality units (polystyrene or poly film for C, metal film for R) for best stability in music circuits. And speaking of this, the 566 has a temperature/frequency drift of around 200ppm/°C, which isn't too shabby. Nonetheless, you'll probably need some additional temperature compensation circuitry if you're designing a keyboard synthesizer VCO.

A control voltage can be applied to pin 5 to vary the output frequency somewhat. Here are several things to keep in mind:

- ♦ The input impedance is quite good, at about 1M. This implies that the input won't load down the driving circuitry appreciably.
- ♦ The response is linear. So, for example (depending on component values), each 0.25V increase might yield a 4000Hz step in frequency at the output. As a rule, a linear response is not of great value for most musical applications but could be ideal in other applications.
- ♦ The control voltage is referenced to V+ and is all kitty-wampus. In particular, as the control voltage *decreases* from V+, the output frequency *increases*.
- ♦ Related to this, according to the spec sheet, the control voltage must lie between 0.75V+ and V+. With a +15V power supply, for example, the control voltage should swing from +11.25V to +15V.
- ♦ As the control voltage comes very close to V+ (giving a low frequency output), the triangle wave becomes increasingly asymmetrical and distorted, until the unit stops oscillating altogether.

Don't let any of this weirdness bother you, for as you'll see it's easy to correct some of the peculiarities of the 566. In the meanwhile, just try to get a feel for the basic functions and connections of the pins. You'll be in better shape to coerce the chip into unusual configurations if you understand what the manufacturer originally had in mind.

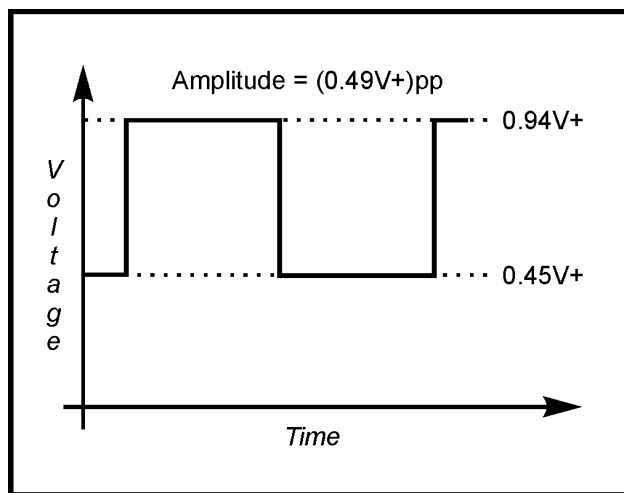


Figure 1.1-2  
Approximate Square Wave Levels

The square wave output is available at pin 3, and the triangle wave at pin 4. Since both of these signals are buffered internally, you can usually just tap them directly for whatever purpose you have in mind. The buffers have a good output impedance of 500

But note that neither signal is referenced to ground or any other convenient voltage level. So, you'll probably want to AC couple them and/or shift them one way or another for most applications.

Figure 1.1-2 is a snapshot of how the square wave behaves. Notice that the baseline is at about 0.45V+ and signal maxes out at around 0.94V+. If the power supply voltage V+ is +15V for example, the square wave amplitude will be near 7.35Vpp, swinging from +6.75V to +14.1V. Please note that these values are approximate and may vary by several tenths of a volt in actual practice.

Figure 1.1-3 shows how the amplitude of the triangle wave depends upon the supply voltage  $V+$ . Observe how much more convenient the constants are in this case as compared to the square wave. I suppose I should mention that I arrived at the formulas for both the square and triangle wave levels simply by measuring them at the breadboard. It is possible that chip-to-chip variation and other factors could give you slightly different results. So, for ultra precise work you might need to offset and amplify the outputs with external op-amps.

And just to wrap up our tour of the features, notice that pin 2 is a do nothing; you can just let it float.

Let's turn our attention next to the frequency of the outputs. We've just seen that there are three factors which determine this: C, R and the control voltage V. A fairly simple formula can be used to compute the approximate frequency based upon these:

$$\text{Frequency} = \frac{k[(V+) - V]}{RC(V+)}$$

$k = 2 \text{ for the NE566}$   
 $k = 2.4 \text{ for the LM566}$

From this we see that the frequency is directly proportional to the negative of the control voltage and inversely proportional to R and C. The supply voltage  $V+$  also factors in. Note the constant  $k$ . The data sheet from Signetics shows this as 2, while National Semiconductor specifies it as 2.4. Take your pick, but in any event the formula should only be used to approximate the frequency of the standard circuit. To keep things simple in what follows, we'll assume the supply voltage  $V+$  is +15V, and  $k = 2.4$  (since you're most apt to run across the LM566).

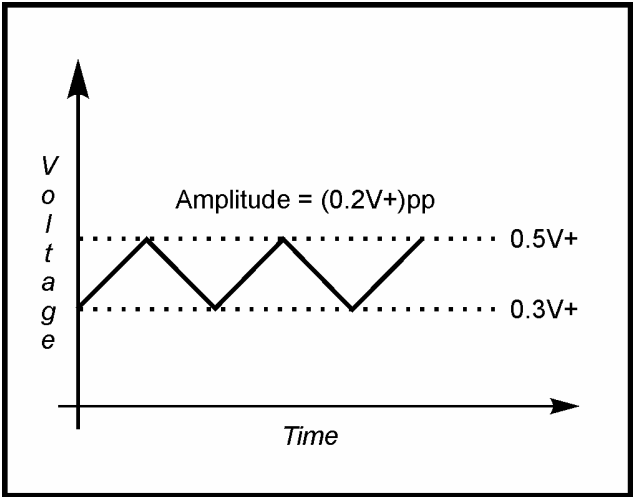


Figure 1.1-3  
Approximate Triangle Wave Levels

V	Hz
11.25	60,000
11.50	56,000
11.75	52,000
12.00	48,000
12.25	44,000
12.50	40,000
12.75	36,000
13.00	32,000
13.25	28,000
13.50	24,000
13.75	20,000
14.00	16,000
14.25	12,000
14.50	8,000
14.75	4,000
15.00	0

Fig. 1.1-4  
Predicted Response

A spreadsheet can be employed to easily create helpful design tables using this formula. For example, let's fix R at 10K and C at 0.001 $\mu$ F. Then, the table in Figure 1.1-4 created with a garden variety spreadsheet program shows how the frequency changes as we vary the control voltage from +11.25V (that's 0.75V+) on up to +15V. The input voltage steps by 0.25V increments, while the output frequency is shown in Hertz. Observe that this is indeed a linear relationship, with a slope of about 4000Hz/0.25V. Further note that in the ideal case, the frequency would eventually reach zero when the control hits +15V, the supply voltage. However, as described earlier, the real world circuit starts to give lousy output waveforms when V gets close to the supply voltage and eventually stops oscillating altogether.

Let's try a more ambitious example. This time we'll fix the control voltage at +13V (with the supply still at +15V), and make a table showing what frequencies to expect for various combinations of R and C. See Figure 1.1-5. Read the values of R along the left vertical axis and the values of C along the top horizontal axis. You might need to count zeroes or translate scientific notation a bit in order to keep the decimals straight. For example, a cap of 1.0E-9 is a 0.001 $\mu$ F one.



<b>R \ C</b>	<b>1.00E-09</b>	<b>2.20E-09</b>	<b>4.70E-09</b>	<b>1.00E-08</b>	<b>2.20E-08</b>	<b>4.70E-08</b>	<b>1.00E-07</b>	<b>2.20E-07</b>	<b>4.70E-07</b>
<b>2,200</b>	145,455	66,116	30,948	14,545	6,612	3,095	1,455	661	309
<b>2,700</b>	118,519	53,872	25,217	11,852	5,387	2,522	1,185	539	252
<b>3,300</b>	96,970	44,077	20,632	9,697	4,408	2,063	970	441	206
<b>3,900</b>	82,051	37,296	17,458	8,205	3,730	1,746	821	373	175
<b>4,700</b>	68,085	30,948	14,486	6,809	3,095	1,449	681	309	145
<b>5,600</b>	57,143	25,974	12,158	5,714	2,597	1,216	571	260	122
<b>6,500</b>	47,059	21,390	10,013	4,706	2,139	1,001	471	214	100
<b>8,200</b>	39,024	17,738	8,303	3,902	1,774	830	390	177	83
<b>10,000</b>	32,000	14,545	6,809	3,200	1,455	681	320	145	68
<b>12,000</b>	26,667	12,121	5,674	2,667	1,212	567	267	121	57
<b>15,000</b>	21,333	9,697	4,539	2,133	970	454	213	97	45
<b>18,000</b>	17,778	8,081	3,783	1,778	808	378	178	81	38
<b>22,000</b>	14,545	6,612	3,095	1,455	661	309	145	66	31

Fig. 1.1-5 - Predicted Response for Some Standard Values of R and C (V held at +13 V)

Just to make sure you know how to interpret the numbers, see if you can figure out what the frequency is when R is 18K and C is 0.022 $\mu$ F. Did you arrive at 808Hz? Anyway, you get the general idea. With a basic spreadsheet program you can readily come up with your own customized tables showing approximately what to expect for various combinations of V+, V, R and C. Electronic design has never been simpler thanks to a program spawned by the financial world!

That pretty much covers the basic operation and arrangement of the 566 VCO chip. Let's pull everything together now and look at a complete, practical circuit. You'll find it in *Figure 1.1-7*, along with a summary of the design notes just covered. We'll recap the essentials just to make sure everything is clear.

V+ is applied to pin 8 and ground to pin 1. Observe also that a 4.7 $\mu$ F electrolytic is strung across these pins; its purpose is to decouple the chip from anything else in the circuit and is essential for good results. Solder it as closely as possible to the power pins.

R and C set the basic timing as mentioned earlier, and they connect to pins 6 and 7 respectively. And the control voltage is applied to pin 5. The resulting output frequency is given approximately by the formula we just hashed over.

The data sheet suggests that pulldown or load resistors should be used on the outputs, and these are shown on pin 3 (square wave) and pin 4 (triangle wave). These could probably be eliminated in some circumstances depending upon the following circuitry.

<b>V</b>	<b>Hz</b>
11.25	59,347
11.50	55,866
11.75	52,356
12.00	48,193
12.25	43,478
12.50	40,000
12.75	35,461
13.00	31,447
13.25	27,624
13.50	23,696
13.75	19,084
14.00	15,152
14.25	10,941
14.50	6,666
14.75	2,392
15.00	Fails

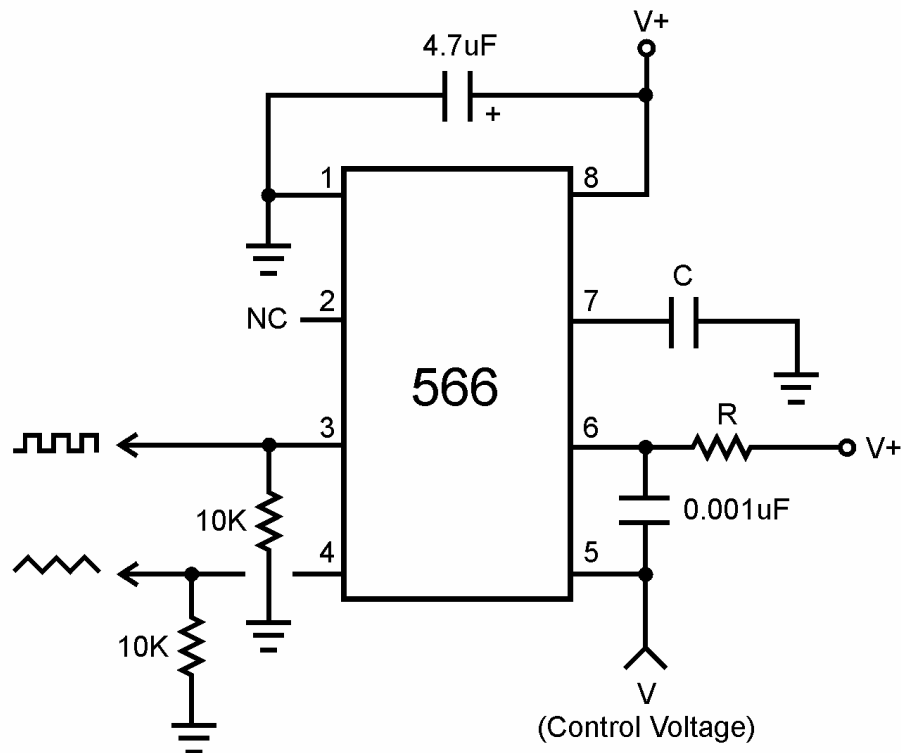
**Fig. 1.1-6**  
Actual Response

Notice how a 0.001 $\mu$ F cap straddles pins 5 and 6; this is used to stabilize the voltage controlled current source and helps to avoid spurious oscillation.

And that's it. I've called this the "standard arrangement," since this is more or less the circuit shown in most data sheets and guide books. The table in *Figure 1.1-6* shows some actual measurements I took at the workbench. I used a +15V supply (actually, the supply weighed in at +14.89V if you want to be precise), with R set at 10K and C at 0.001 $\mu$ F. I guesstimated the frequencies on an oscilloscope, so a few inaccuracies may have crept in here and there. Nonetheless, compare this with the predicted response in *Figure 1.1-5* and you'll see that the formula is surprisingly good, especially in its midrange.

By the time the control voltage V hit +14.75V I could just barely detect the waveform distortion mentioned earlier; the triangle wave started to slump ever so slightly. Over the last quarter volt, as I approached +15V, it got worse and worse, and then the unit fizzled to a standstill.

The configuration does work well and is reliable, but as mentioned several times before, is somewhat limited. Mainly, the linear 1:10 sweep is a bit anemic, and we sure could use other outputs than just a simple square or triangle. Well, not to worry-in the remainder of this guide I'll show you how to go way beyond what the manufacturer thought was possible. So, if you've got the standard arrangement down pat, then turn some pages and see how to put the 566 on steroids!



### Design Notes

- The supply voltage  $V+$  can be from +12V to +24V.
- The current drain is about 10mA.
- Timing resistor  $R$  must be between 2K and 20K.
- The corresponding current into pin 6 must be between 100uA and 1000uA.
- The control voltage  $V$  must be between  $0.75V+$  and  $V+$ . Note that the control voltage on pin 5 is referenced to  $V+$ , hence an increase in voltage yields a decrease in frequency.
- The impedance of the control voltage input at pin 5 is a nominal 1M.
- The output frequency is given approximately by the formula:

$$\text{frequency} = \frac{k[(V+) - V]}{RC(V+)}$$

$k = 2$  for the NE566  
 $k = 2.4$  for the LM566

- Maximum guaranteed frequency is 1MHz.
- The 0.001uF capacitor straddling pins 5 and 6 is required to eliminate possible oscillation in the voltage controlled current source within the 566. The 4.7uF electrolytic on the supply lines is employed to prevent the 566 from interacting with any other subcircuits. Connect it as closely as possible to pins 1 and 8.

*Figure 1.1-7  
Standard Arrangement for a VCO with a 1:10 Linear Sweep*

## 1.2 Wide Range Linear VCO

As promised, here is an incredible improvement on the basic linear VCO design. The manufacturer (and most other nay-sayers) will tell you that the sweep range of the 566 is constrained to a measly 1:10. Would you believe we can increase this by a factor of 100, yielding an astounding 1:1000 sweep?

I first saw this magic in an article by the amazingly creative John Simonton, founder of PAiA Electronics. (I think John is one of the cleverest designers around. And he's also one of the nicest guys you'll ever meet.) The bibliographic specifics are, "Potpourri and the Apple Connection," *Polyphony*, November 1977, pp. 28 - 31. As the title of his article suggests, John covered a number of disparate topics here, and the 566 thing was almost lost in the mix. I've never heard anyone else mention his discovery, nor have I seen it in print elsewhere; this may be the best kept secret in electronic music!

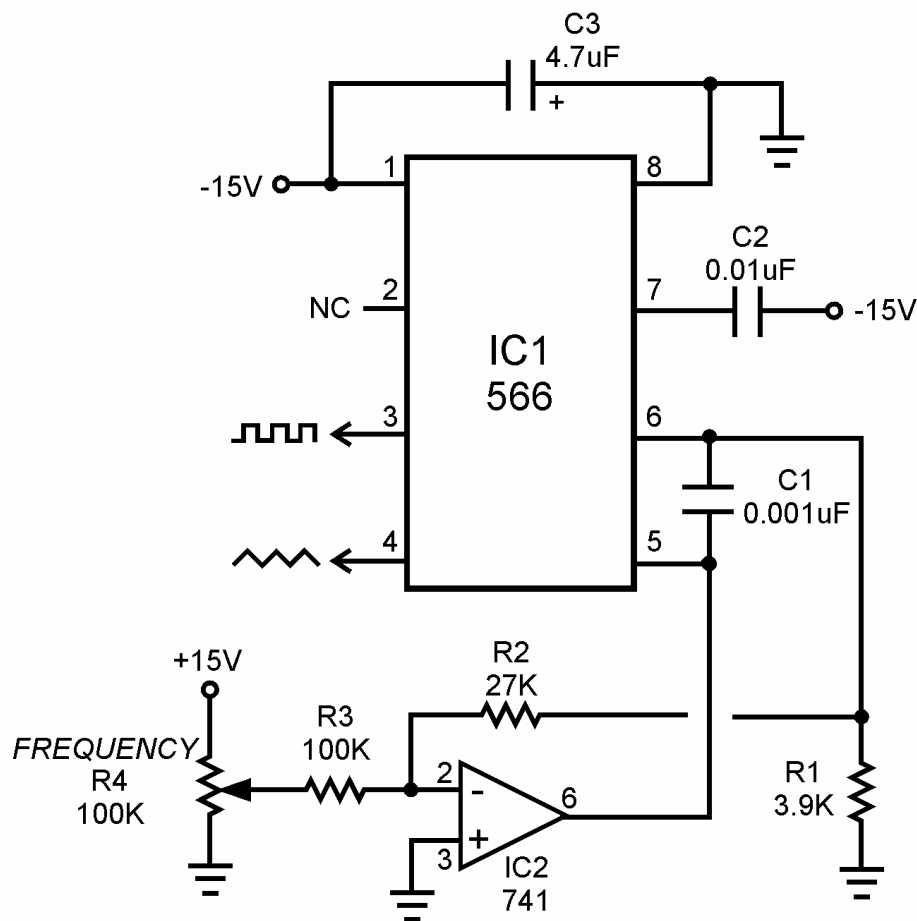
John gave the basic circuit topology in the article mentioned above, but didn't specify any component values. Through careful experimentation I've been able to arrive at some suitable ones, and have also simplified his design a bit for our purposes. The general idea is that the voltage controlled current source in the 566 can be greatly improved by putting it within the feedback loop of an op-amp. In loose terms, the benefit here is that the source is now more-or-less unloaded, and so the linear response is extended. In particular, the performance is much more predictable, especially at the two extremes of high and low control voltage inputs.

But wait, there's more (as they say on TV ads). By putting the 566's current source in the *negative* feedback loop of an op-amp, the unit now responds *directly* to the control voltage input. That is, an increase in voltage yields an increase in output frequency. And best of all, by powering the 566 off of a negative supply, the control voltage is now referenced to ground or 0V, making things even more pleasant. Like I say, this is sorcery!

Refer to *Figure 1.2-1* for the details. The first thing to note is the unusual power supply configuration. Here a -15V is applied to pin 1 and ground to pin 8. The 566 doesn't really care, since as far as it's concerned, there's still a +15V potential straddling the two pins. But, as mentioned, this has the effect of referencing the control voltage input at pin 5 to ground—a much more convenient scenario.

V	Hz	
17mV	11	If you chase the op-amp connections around you will note that the current source (between pins 5 and 6 of the 566) is indeed now in the negative feedback loop. C1 is still used, however, to stabilize the internal source. C2 is the timing capacitor, and you'll see that it now connects to the negative supply instead of ground.
31mV	38	
0.25	345	
0.50	741	
0.75	952	
1.00	1,333	I've shown a simple pot, R4, as providing the control voltage input. But really, this can be any voltage source you desire, like the output of an envelope generator, an LFO and so on. I took some measurements at the workbench (again, estimating the frequency on an oscilloscope) and <i>Figure 1.2-1</i> shows the results I observed.
2.00	2,597	
3.00	3,846	
4.00	5,128	
5.00	6,309	
6.00	7,634	From +1V to almost +15V (recall that my bench supply actually tops out at +14.89V on the positive side), the response is quite linear with a slope of about 1220Hz per volt on average. The waveforms are excellent throughout this range. By the time the control voltage drops to +0.25V I can just begin to perceive a bit of distortion, but not enough to really be objectionable. As the frequency falls to 38Hz and further to 11Hz, the symmetry of the output changes substantially, but otherwise the 566 keeps oscillating reliably. That's a sweep range of well over 1:1000!
7.00	8,772	
8.00	10,000	
9.00	11,236	
10.00	12,500	
11.00	13,514	By the way, the waveform becomes distorted only at the extreme low end due to imperfections in the triangle buffer within the chip. John Simonton proposed a fix for this in his article mentioned above, but I was never able to get much out of it. As it is, the circuit performs remarkably well.
12.00	14,793	
13.00	16,026	
14.00	17,241	
14.89	18,182	

**Fig. 1.2-1**  
*Actual Response*



### Design Notes

- Note the unusual power supply connections to the 566. This has the benefit of now referencing the control voltage on pin 5 to ground (instead of V+).
- With the values shown, this circuit has a range of at least 18Hz to 18KHz, a 1:1000 ratio! The response is approximately linear. That is, for each 1V increase of potentiometer R4, the output frequency will increase by about 1220Hz.
- With the values shown, the current into pin 6 of IC1 attains a maximum value of 1000uA, as required by the spec sheet. The voltage across pins 5 and 1 attains a minimum of about +11.25V which is 0.75V+ (also mentioned in the spec sheet).
- As the frequency drops below 400Hz or so, the triangle wave becomes increasingly distorted, slumping into a ramp. Still lower, corresponding to a tiny control voltage input, the 566 stops oscillating altogether. To avoid the lower range, you may feed in a fixed offset (set by a trimmer through a resistor) to pin 2 of IC2. Alternately, you could put in an optional trimmer in series with potentiometer R4, between the bottom lug and ground. This will prevent the control voltage from dropping below a small fixed value.
- For low frequency operation, increase timing capacitor C2. For example, doubling the value of this capacitor will halve the frequency.

Figure 1.2-2  
Linear VCO with a 1:1000 Sweep Range

## 1.3 Exponential VCO

---

The circuit of the last section considerably extends the usable range of the 566. It lets you alter its output frequency reliably over a ratio of 1: 1000, from around 18Hz on up to 18KHz. But try to imagine what it might *feel* like if you were using a potentiometer to sweep the frequency. Due to the linear nature of the circuit (and the fact that musical pitch is exponential in nature, doubling in frequency for each new octave), you'd hear a lot of change when you just started to twiddle the dial. But as you continued to move upward, you'd find that you would have to keep turning it further and further to attain the next octave. In other words, the scale of change on the pot would be cramped at the start and extremely spread out during the final three-fourths of its rotation. To put it more succinctly, exponential control is perhaps more musically useful than linear control.

Most people aren't aware of it, but it's actually quite easy to add exponential control to the 566 VCO. In fact, with just a few extra components it is possible to come very close to a true 1 V/octave scale. As mentioned earlier, a rock-solid VCO that doesn't shift with temperature is typically fairly complex and sophisticated. Nonetheless, for other situations (like electronic percussion) the wide range and musical "feel" of the simple exponential VCO shown here might be just the ticket. And the circuit is hardly more complicated than its linear cousin.

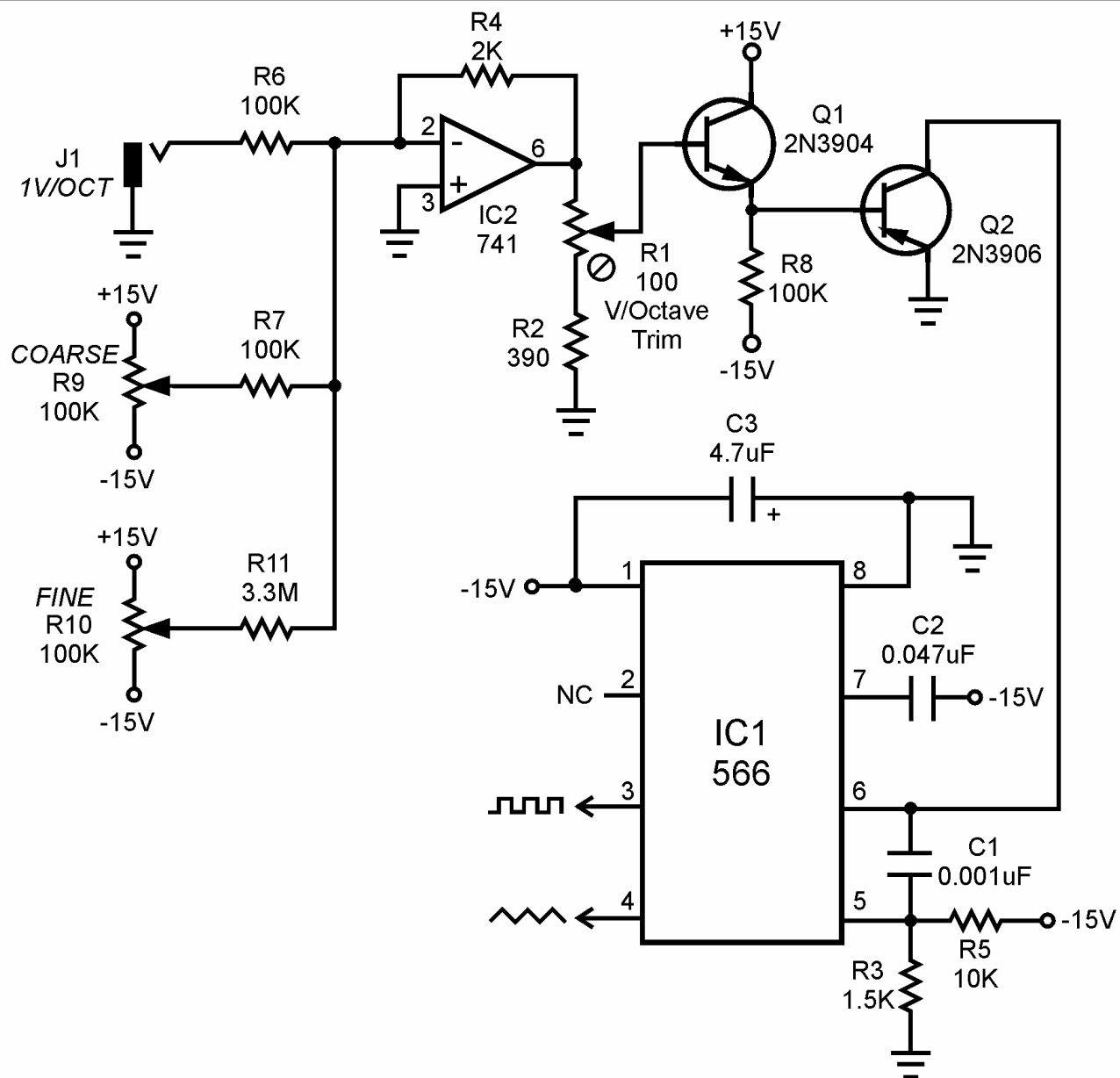
Pin 6 of the 566 is actually a current sink. If we can come up with a current source to attach to it which is exponential in nature, then we're home free. In fact, Terry Mikulic showed how to design just such an exponential current source in his article, "Exponential Converters," *Electronotes*, Volume 5, Number 37, March 30, 1975. We'll use a simplified version of his clever idea in the circuit here.

Refer to *Figure 1.3-1* which shows the complete schematic for an exponential VCO using the 566 chip. The converter consists of Q1 and Q2. Q1 is configured as a simple emitter follower, while Q2 performs the voltage to-current conversion. The basic idea is that the collector current in Q2 is an exponential function of the base-emitter voltage. But if you chase through the mathematics of the relationship and consider how a typical transistor behaves, you'll find that the emitter saturation current is extremely sensitive to temperature changes. This is no good. But Q1, besides acting as a buffer, also responds to temperature—but in the reverse direction. If Q1 and Q2 are well matched for the emitter saturation characteristic, then the errors cancel out and the converter remains stable with respect to temperature. If the matching is perfect, then this would be an ideal VCO for the most demanding circumstances. However, it is difficult for manufacturers to match NPN and PNP transistors to this degree, so we'll accept that using some garden variety units like the 2N3904 and 2N3906 gives acceptable results for many applications, just not the stability required of a keyboard synth.

### An Interesting experiment

If you breadboard the circuit in Figure 9, try the following: Monitor the output of the VCO through an amplifier/speaker combination. Now touch one of the transistors with a finger. You'll hear the pitch increase in frequency as heat is transferred from your body to the component. Then touch the other transistor and you'll hear the pitch drop back down. Theoretically, if the two transistors are well matched and in close contact with each other, the current changes should cancel each other out.

Now that we have the basic idea under our belts, let's wrap up the details. Again, refer to *Figure 1.3-1*. IC2 forms the control voltage summer. I've shown the circuit with three inputs: a coarse tuning control, a fine tuning control and a 1 V/octave input. You'll note that R10 is a very large 3.3M, which is why the fine control has less of an effect than the coarse control. Incidentally, you may or may not need all three of these inputs, depending on what you're designing. In fact, for many circuits, a single control pot (like R8) might be all that is needed. If that's what you end up doing, then you may want to tie a fixed resistor on each end of R8 to keep the VCO away from any possible dead spots at the two extremes.



## Design Notes

- Again, note the unusual power supply connections to the 566.
- The control input at pin 6 of IC1 is a current sink which is compatible with the current source of the Q1/Q2 transistor pair.
- For good temperature stability, Q1 and Q2 should be epoxied together.
- For even better temperature stability, R4 should be a 2K, +3300ppm/°C thermistor, and mounted in close contact with Q1 and Q2.
- The response is about 1 volt per octave. See the text for details on the actual usable range.

Figure 1.3-1  
Exponential VCO

The op-amp and related components scale the control voltage appropriately. With the values shown, it is possible to tweak trimmer R1 to get a very decent 1V/Octave response. Note resistor R4 in the feedback loop of the op-amp. If you make this a +3300ppm/°C thermistor (once very common in analog synth circuits), then you can cancel out the remaining temperature dependence in the scale factor. If you decide to go this route, then Q1, Q2 and R4 should be epoxied together so that they all ride at the same temperature. However for most control or drum circuits, you can simply use an ordinary resistor for R4 (but I would still glue Q1 and Q2 together since it's so easy to do).

Notice once again that we power the 566 chip from the negative supply, applying -15V to pin 1 and ground to pin 8. R3 and R5 bias the control voltage input at pin 5 to about -2V. But thanks to the reversed nature of the power supply, you can think of this as a potential of +13 V on pin 5 with respect to pin 1. This puts it in the middle of the desirable range (between 0.75V+ and V+) as described in Section 1.1.

To give you a feel for how well this circuit performs let me mention that I built it up as shown and controlled it with a precise 1 V/Octave keyboard. (The actual details are that I used a Casio CZ-1000 digital synth, whose MIDI output was connected to the Midwest Analog Products MTS-100 Midi-to-Synth converter module). I was able to tweak up R1 quite nicely and achieve excellent tuning over the middle five or six octaves. In this case, I was using my ears as the test instrument and the results were impressive for such a simple circuit. But to be fair, I should also mention that if I changed the temperature of the VCO by breathing on it or touching the transistors, I could easily detune the unit.

V	Hz	
-6.00	10	But you'd probably like some numbers so you can see how large a range the circuit has. I monitored the middle lug of R8 with a digital multimeter and estimated the output frequency on an oscilloscope. (I don't own or usually need a frequency counter for most of my work). The results I tallied are presented in <i>Figure 1.3-2</i> .
-5.00	22	
-4.00	43	
-3.00	83	
-2.00	167	
-1.00	345	There are several things to note. First, the scale is indeed exponential in nature (base 2), with each increase of 1V at the control input leading to an approximate doubling in output frequency. As I mentioned, when using ears as the final arbiter, the tuning was quite respectable at frequencies below 5KHz or so.
0.00	690	
1.00	1,401	
2.00	2,703	
3.00	5,236	
4.00	9,436	Secondly, observe how the output begins to go noticeably flat once you get up around 9KHz. This is typical in circuits of this type when the control current becomes larger and larger. The culprit is something called the "bulk emitter resistance" occurring in Q2. There are ways of correcting this by injecting a non-linear offset into the converter. The general idea is that an extra voltage is added to the input, but only at the higher end where the "going flat" phenomenon occurs. In fact, Terry Mikulic showed how to easily accomplish that in his article mentioned above. It only takes a handful of other components, but due to the inherent problem of finding matched NPN and PNP transistors, as described earlier, I've deemed it not worth pursuing for this circuit. Like I say, if ultra precision is your goal, then there are more effective ways of designing an exponential VCO for synth work.
5.00	15,385	
6.00	20,408	
7.00	23,585	
8.00	23,810	

**Fig. 1.3-2**  
*Actual Response*

Finally, I detected on the oscilloscope that at a -3V control voltage the triangle wave just began to show the traditional slump which occurs on the low end (for review, recall that this is due to imperfections of the 566's internal buffers). At -4V and even -5V the distortion was still very reasonable. At -6V however, corresponding to a frequency of 10Hz, the waveform had degenerated into a very noticeable ramp. But even if you consider 22Hz to be the lowest usable frequency, observe that this circuit still boasts a robust 1:1000 sweep range - that's a 10 octave span! Altogether it is a remarkably useful addition to the electronic music designer's arsenal.

## 1.4 Triangle and Sine Wave Outputs

---

In this and the next several sections we turn our attention to the outputs of 566 VCO circuits. As it comes from the factory, the 566 provides only triangle and square wave outputs, and even these are a somewhat nonstandard in value. But with just bit of creativity we can fix that and also greatly expand our options. Let's start with the triangle wave.

As mentioned in Section 1.1, the triangle output at pin 4 is a little weird. The center line is biased away from ground, and the amplitude is a trifle weak. Fortunately, both of these deficits are easily corrected. Refer to *Figure 1.4-1*. To take care of the DC offset, we simply use a coupling capacitor, C1. If you're primarily interested in audio frequencies (20Hz to 20KHz), then a value of 0.22 $\mu$ F is more than sufficient. For lower frequencies you can increase this, even using an electrolytic as long as you're careful to obey the polarity. For example, if you're exploiting the standard arrangement of Section 1.1 with a positive power supply, then the plus side of the cap would connect to pin 4 of the 566. If using a negative supply as in Sections 1.2 and 1.3, then the negative side of C1 would connect to pin 4. .

If you're designing a super low frequency LFO, then you'll probably want to discard the cap altogether and go with a direct coupled affair. In this case you'll need to bias the triangle wave by summing an offset into IC1a. Any elementary op-amp book will guide you through this.

But back to the problem at hand. While C1 takes care of the DC bias, op-amp IC1a scales the wave up to a more convenient amplitude. The ratio of resistors R14 and R11 set the gain at 10/3. Assuming you're using a 15V power supply (as all of the circuits in this guide employ), then the triangle wave at pin 4 will have an amplitude of 3Vpp. Multiply this by 10/3 and you get 10Vpp. This is a standard value for virtually all analog synth gear and that's what we'll shoot for throughout. By the way, notice that the output impedance at jack J1 is 1K, another synth standard we'll adhere to.

Let's consider the sine wave output now. There are essentially two non-reactive ways to convert a triangle wave to a sine wave: stepwise linear approximations and non-linear transform circuits. I've never liked the former method, considering it just too awkward and complicated for the "stick-like" results it offers. (Stepwise linear approximations are used in some of the older PAiA synthesizer modules and in the 8038 VCO chip, though). With regard to the latter method of non-linear circuits, I've played with many approaches over the past quarter century. Some exploit the characteristics of FET transistors, while others go for a full chip design.

But in my estimation, the best method is that utilizing a couple of discrete transistors wired as a differential pair. The idea dates back almost 40 years, when the article "Non-Reactive Filter Converts Triangle Waves to Sines" appeared in *Electronics*, March 8, 1965; the authors were R. D. Middlebrook and F. Richer. This approach really caught on and by the late seventies, versions of it showed up in all sorts of commercial and homebrew synth gear.

With the ready availability and low price of integrated circuits, later designers set out to "improve" the idea by utilizing the differential pair within the 3080 transconductance op-amp, mimicking the original concept. But to my ears, this was a step backwards. I believe the earlier discrete version sounds better, is simpler and is cheaper to build. Best of all, it is possible to attain distortion levels well below 1%.

The mathematics of how the circuit works is fascinating. Should you have a turn for that sort of mental activity (like I do), then check out the sources in the sidebar on the next page. But for now, we'll skip over the calculations and instead concentrate on the practical details. As I mentioned earlier, I've been playing with the differential pair notion for many years and have arrived at some component values that really give superior results. And there is one additional trick I'll turn you on to in just a moment that propels this circuit to the head of the class.



## Technical Sources on the Differential Pair Triangle to Sine Converter

W. A. Evans and V. Schiffer, "A Low Distortion Triangle to Sine Converter," *The Radio and Electronic Engineer*, Volume 47, Number 5, pp. 217 - 224.

Bernie Hutchins, *The Musical Engineer's Handbook*, (Ithaca, New York: Electronotes, 1975), pp. 5B-18 through 5B-21.

Walter G. Jung, *IC Array Cookbook*, (Rochelle Park, New Jersey: Hayden Book Company, 1980), pp. 121, 122.

Robert G. Meyer, Willy M. C. Sansen, Sik Lui, and Stefan Peeters, "The Differential Pair as a Triangle-Sine Wave Converter," *IEEE Journal of Solid-State Circuits*, June 1974, pp. 418 - 420.

Continue to refer to *Figure 1.4-1*. Recall that we've already sized up the triangle wave to be 10Vpp. This happens to be the correct value needed by the input of the triangle-to-sine converter. We'll tap into it and apply it to the base of Q1, which along with Q2 forms a differential pair. We deliberately overdrive the base of this transistor forcing it into its nonlinear region. The funny thing is, when overdriven like this, the resulting signal begins to uncannily approximate a sine wave. Trimmer R13 is used to adjust the "roundness" of the resulting sine approximation. Tweak it in one direction and the wave gets pointy like the original triangle; go too far in the other direction and it flattens out like a rectangle. But when you hit the sweet spot in between, the output indeed curves in a pleasant manner and assumes its new guise as a sine impostor.

The second trimmer, R12, let's you adjust the symmetry of the approximation. Go too far to the left and the output will be skinny and pointed on one side while squat on the other. Spin R12 the other way, and the reverse happens. At the correct setting, however, you'll get a neat, symmetrical waveform. In terms of what your ears can hear, at one extreme or the other you'll detect some extraneous harmonics giving a rather ragged sound. At the correct setting, though, you'll hear a pure, flute like tone.

Moving on, the outputs are tapped at the two collectors and fed to op-amp IC1b, which is configured as a differential amplifier. With the values of all the resistors as shown, the sine approximation appearing at J2 is a nominal 10Vpp, with a 1K output impedance. If you'd like to adjust the output amplitude one way or another, simply alter R8 and R9. To maintain a DC center-line of 0V, these two must be balanced and so should be changed by an identical amount. A larger value gives a heftier output and a smaller value yields a lesser one.

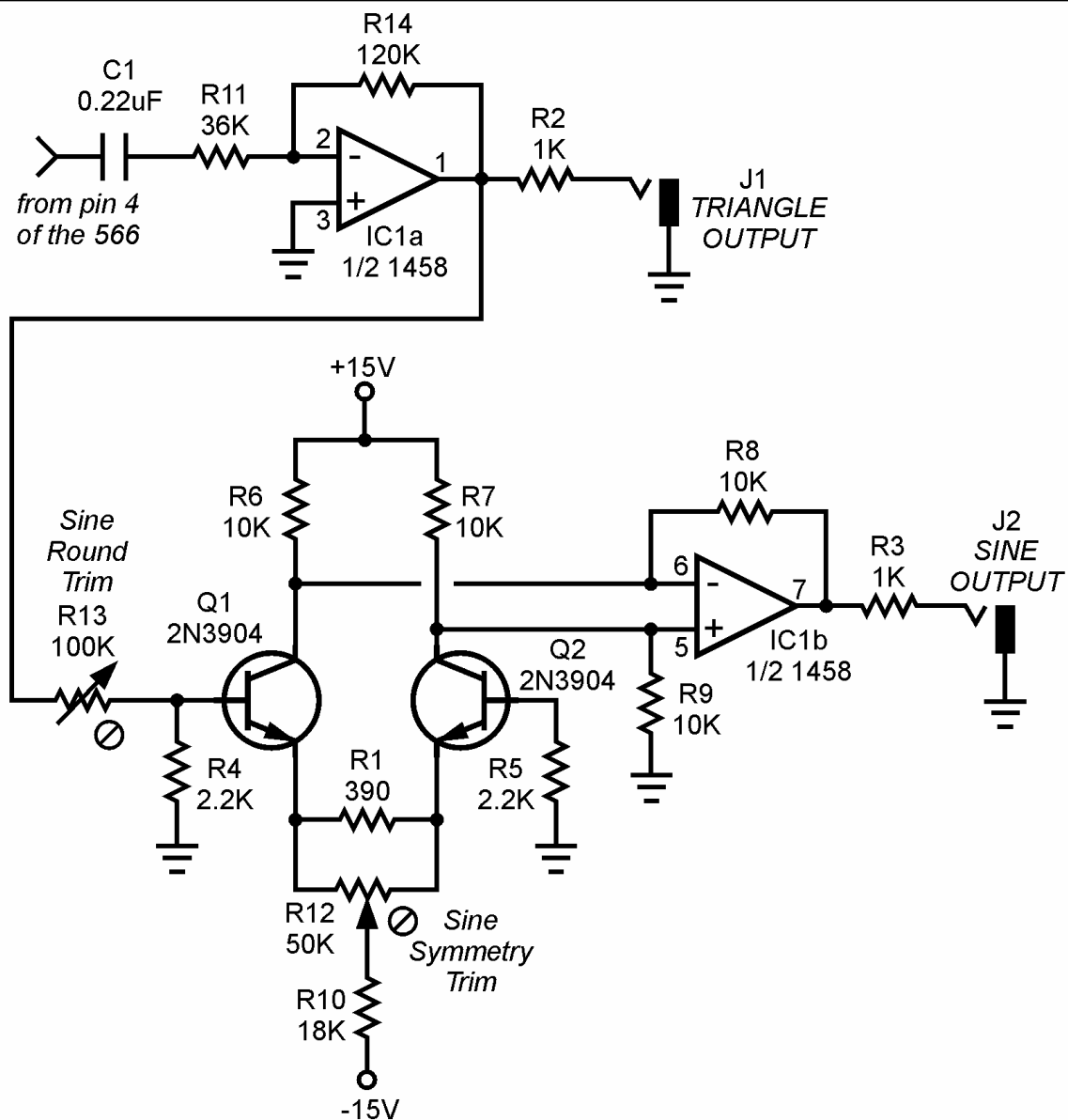
And now, about that secret ingredient I alluded to. It's nothing more than the single resistor R1. While looking fairly innocuous, it in fact provides a feedback path within the differential pair. This has the effect of practically nulling out the pip which can occur at the extreme high and low points of the sine wave approximation. The result is not only a great looking sine wave suitable for driving lasers or other light show equipment, but a waveform that sounds completely free of harmonic content in musical instruments.

Incidentally, I believe the noted synthesizer designer Denis Colin used a discrete differential pair in his classic Aries 317 VCO. But according to my notes he employed a feedforward correction, not the feedback version I've shown here.

To wrap things up, I should probably say something about how to tweak the trimmers. One approach is to connect the output to an oscilloscope. Tune the frequency to about 1KHz for easy viewing. Now simply tweak the pots, one after another for the best looking waveform. This will require several passes "around the horn" to get both of the pots to their optimal positions.

An even better method is to connect the output to an audio amplifier and let your hearing do all of the work. Believe it or not, even the untrained ear can perceive an extremely tiny amount of distortion. Watch the output level, however. Remember this is a hot circuit and you won't want to blow up your ears or your loudspeakers. Again, tweak the pots until no overtones or harmonics are obvious.

Of course, you could always use a harmonic distortion analyzer, but who has one of these lying around? Anyway, your ears (and the ears of your audience) are the final arbiters, so why not give them the job!



### Design Notes

- The driving triangle from pin 4 of the 566 is assumed to be 3Vpp in amplitude, which is what you'll get if using a 15V supply.
- For extended low frequency response, C1 may be replaced with a larger value electrolytic type, but watch the polarity.
- Both the triangle and sine wave outputs (at jacks J1 and J2, respectively) are a nominal 10Vpp in amplitude, with a standard output impedance of 1K.
- Adjust trimmers R12 and R13 for lowest distortion.
- Alter R8 and R9 simultaneously if desired, to adjust the sine wave amplitude.

*Figure 1.4-1  
Triangle and Sine Wave Outputs*

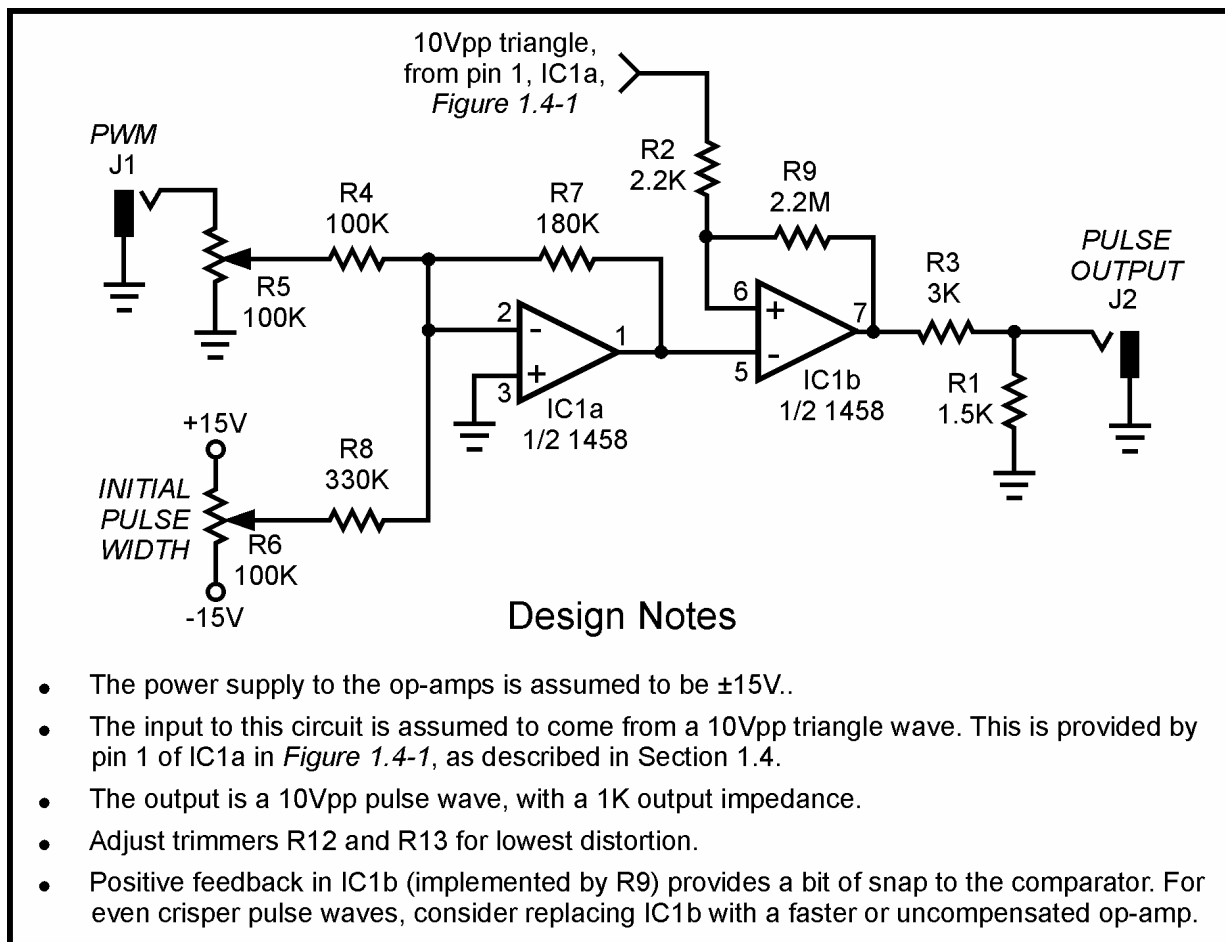
## 1.5 Pulse Width Modulated Output

I'll confess that pulse width modulation is one of my favorite electronic music effects; I love the ethereal "whooshing" sound it produces. And yet a circuit to generate this is among the simplest to implement. The schematic in *Figure 1.5-1* shows how to add this feature to a 566 type VCO.

The heart of the circuit is op-amp IC1b which is configured as a comparator. A 10Vpp triangle wave (available from the circuit described in Section 1.4) is fed into the non-inverting input by means of resistor R2. The instantaneous level of the triangle is "compared" to a fixed DC voltage applied to pin 5. If the level is higher, then the output at pin 7 jumps to near V+. If lower, then it goes to V-. R9 provides a smattering of positive feedback, forcing the output to snap faster and more cleanly than it would otherwise.

Voltage divider R1/R3 chops the output down to the usual 10Vpp level, while simultaneously providing a 1K output impedance. Recall that these are the standards employed by most analog synthesizer gear.

You'll typically want to modulate the pulse width with an LFO, envelope generator or some other control voltage source. IC1a sums a PWM (pulse width modulation) input from J1 with some fixed offset provided by R6. By playing with controls R5 and R6 it is possible to achieve 0% to 100% control over the width of the output rectangular wave.



*Figure 1.5-1*  
*Pulse Width Modulated Output*

## 1.6 Square Wave and Sub-Octave Outputs

As we saw earlier in this guide, the square wave output at pin 3 is not centered about ground and is a trifle weak in amplitude. But it's easy enough to fix these deficits. We'll even toss in a sub-octave output for creating some nice fat bass sounds.

Refer to the circuit in *Figure 1.6-2*. The first thing to keep in mind is that this circuitry has been designed to work with the 566 when it's powered by -15V. Recall that the implementations of Sections 1.2 and 1.3 take advantage of this clever scheme.

The square wave is tapped at pin 3 of the 566, and applied to the comparator comprising IC1a and associated components. Resistor R13 provides a bit of positive feedback to give the comparator a little extra snap. Voltage divider R9/R10 sets a threshold of -7.5V, which works out pretty well for the signal coming from the 566.

The unadorned output at pin 1 of IC1a swings about  $\pm 15V$ , so we chop it down to size with voltage divider R6/R2. The square wave at jack J1 now has an amplitude of about  $\pm 5V$ , and the output impedance is 1K. Recall that these are the synthesizer standards we've been observing throughout this handbook.

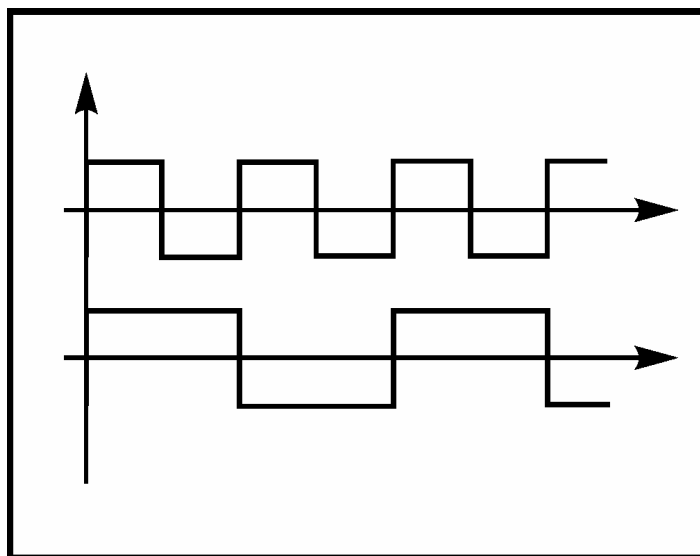
The sub-octave square wave is generated by means of a divide-by-two circuit. This is implemented with a CMOS 4013 flip-flop. The divide by two action is created in the usual manner by cross-coupling the *not-Q* output back to the *D* input (pins 2 and 5, respectively). Note that all of the unused inputs are grounded, which is a standard rule to obey when using CMOS chips.

The square wave output from IC1a is applied to the clock or C input of the 4013. To keep from destroying the chip, diode D1 blocks any negative swings. R11 is simply a pull-down resistor which stabilizes the diode somewhat and also makes the CMOS IC happy.

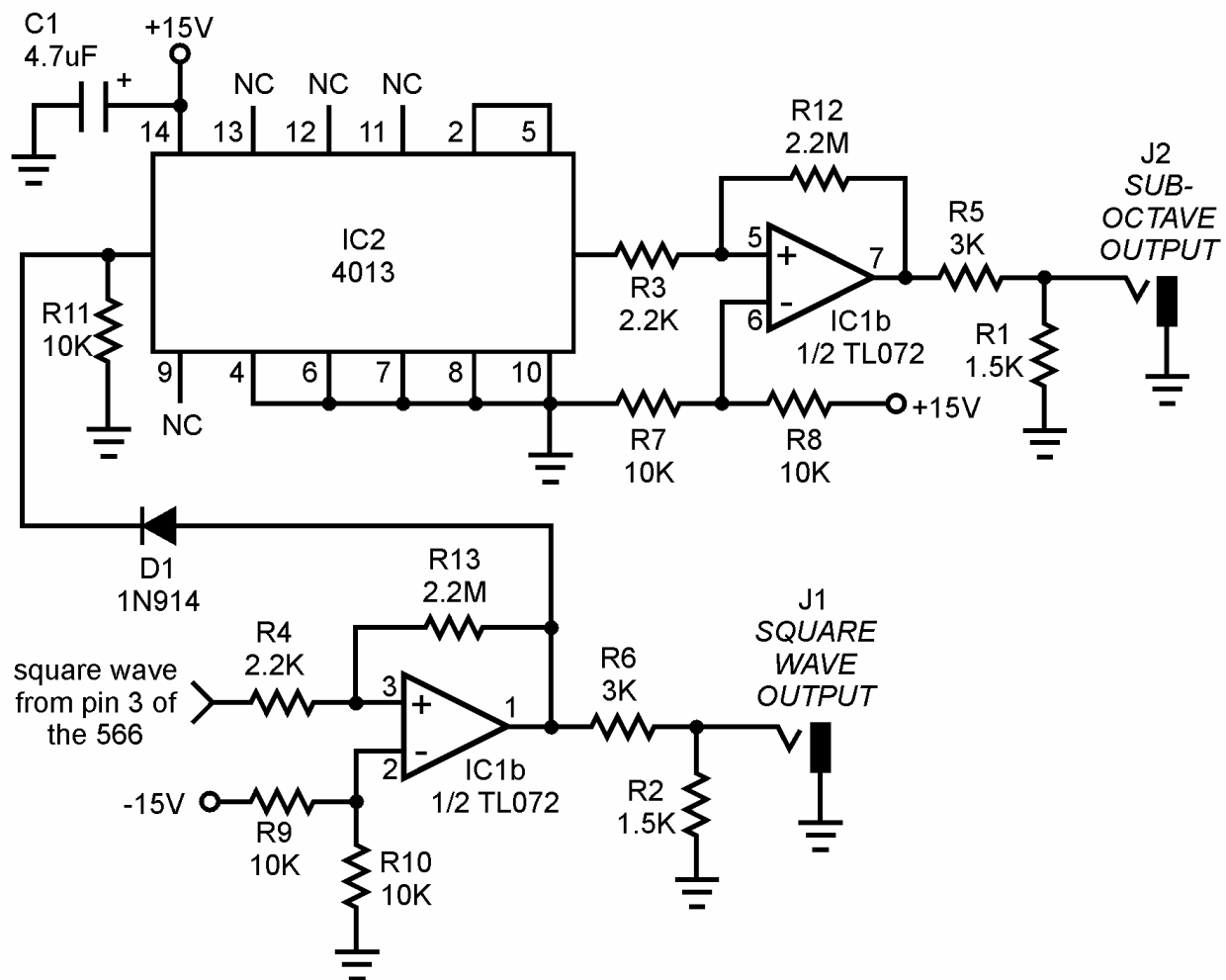
The signal at the output, pin 1 of IC2, is a good looking square, but now one-half the original frequency. See *Figure 1.6-2*. It swings from ground to +15V and so has to be shifted and scaled somewhat. You could do this with a standard mixer/amp combination, but it's easier here just to process the signal with yet another comparator. IC1b is set up in almost the same way as the comparator in the square wave portion. Note, however, that the threshold is now fixed at +7.5V (by means of voltage divider R7/R8) since the 4013 is running off of the positive supply.

I mentioned some unused pins above. The 4013 is in fact a dual unit, containing two independent flip-flops. If you really wanted, you could use the second one to derive a sub-sub-octave output. That ought to give your bass woofers a workout!

Finally, let me mention that I've used the common TL072 dual op-amp to implement the comparators. This is a pretty fast chip, all things considered, and so the outputs are plenty good enough for musical and audio applications. But if you have some sort of digital clocking operation in mind, you may want to replace it with a faster chip, or even an uncompensated type.



*Figure 1.6-1*  
*Square and Sub-Octave Waves*



### Design Notes

- This design assumes that you are powering the 566 from -15V as in the circuits of Sections 1.2 and 1.3
- The square wave from pin 3 of the 566 is applied to R4.
- Pay particular attention to the orientation of D1 which blocks the negative swings of the square wave coming in from IC1a.
- The outputs are both a nominal 50% duty cycle, 10Vpp in amplitude and 1K in impedance.
- For even crisper outputs, IC1a and IC1b could be replaced with faster op-amps or uncompensated comparator types.

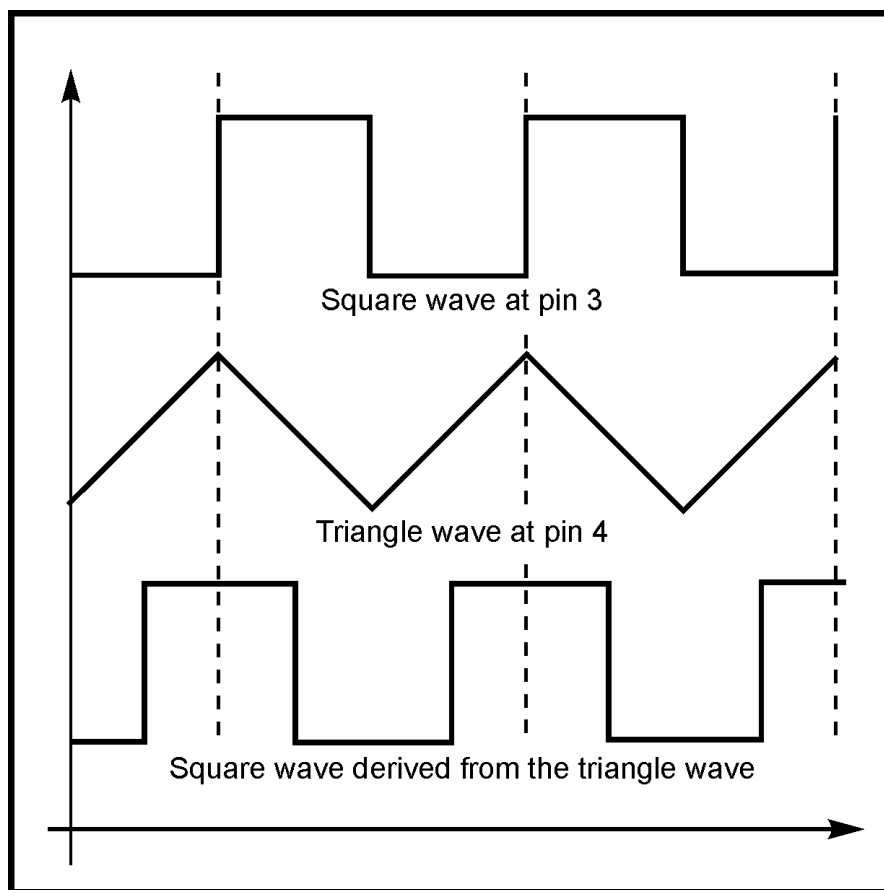
*Figure 1.6-2  
Square Wave and Sub-Octave Outputs*

## 1.7 Quadrature Square Wave Outputs

For most audio and music applications, it's not the absolute phase of the waveform that matters so much, but whether it's changing or not. The human ear is relatively insensitive to phase, but *is* able to detect changes in phase. Nonetheless, in certain digital or control applications (like modulating the choppers on a laser used in a light show) the ability to come up with a couple signals that are precisely separated from one another by a certain phase angle is of great utility. Here's just such a circuit.

In particular, the 566 can be coerced to produce square waves separated by  $90^\circ$ . This might not sound like much to the uninitiated, but in fact it's quite miraculous. To create two squares  $180^\circ$  out of phase is easy - just use an inverting amp. But to generate a shift that is half of that (and non-reactively) is quite a different kettle of fish.

But the 566 makes it a snap. That's because of the way the internal modules are arranged. To understand how, examine *Figure 1.7-1*. The waveform at the top appears at pin 3. Below that is the triangle showing up at pin 4. Notice how these signals are already shifted apart from one another by  $90^\circ$ . If we can convert the triangle to a square, then we're home free.

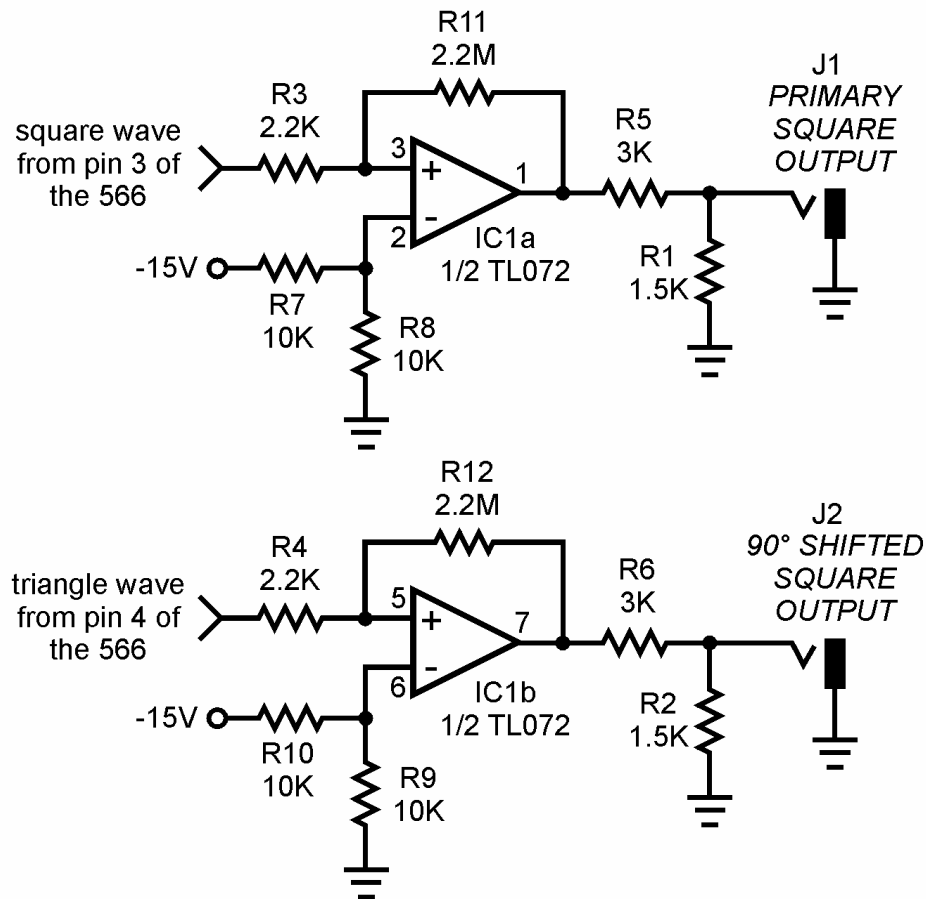


*Figure 1.7-1*  
*Waveform Phase Relationships*

And *that* is easy to do. In particular, we'll design a comparator that changes state when the triangle wave is at its midpoint; the result is a square wave  $90^\circ$  out of sync with the signal at pin 3. Compare the bottom waveform with the triangle and square above it and you'll start to get the picture.

Refer to *Figure 1.7-2* now. The square coming from pin 3 is adjusted to a standard level and impedance using the same circuitry we saw in Section 1.6. As usual, we're assuming you're running the 566 off of a negative power supply.

The triangle wave at pin 4 is also tapped and sent to a similar comparator. Voltage divider R9/R10 sets a threshold of -6V. And it just so happens (see Section 1.1 for proof) that's exactly where the mid level of the triangle wave occurs. Thus the resulting waveform is about 50% in duty cycle - more or less square - and  $90^\circ$  out of phase with the primary signal. Toss in a couple more inverters and you can have all four cardinal angles at your disposal:  $0^\circ$ ,  $90^\circ$ ,  $180^\circ$  and  $270^\circ$ .



### Design Notes

- This circuit assumes that you are powering the 566 from -15V as in the circuits of Sections 1.2 and 1.3.
- The square wave from pin 3 of the 566 is applied to R3.
- The triangle wave from pin 4 of the 566 is applied to R4.
- The outputs are both a nominal 50% duty cycle, 10Vpp in amplitude and 1K in impedance.
- For even crisper outputs, IC1a and IC1b could be replaced with faster op-amps or uncompensated comparator types.

*Figure 1.7-2*  
*A Circuit to Generate Quadrature Square Wave Outputs*

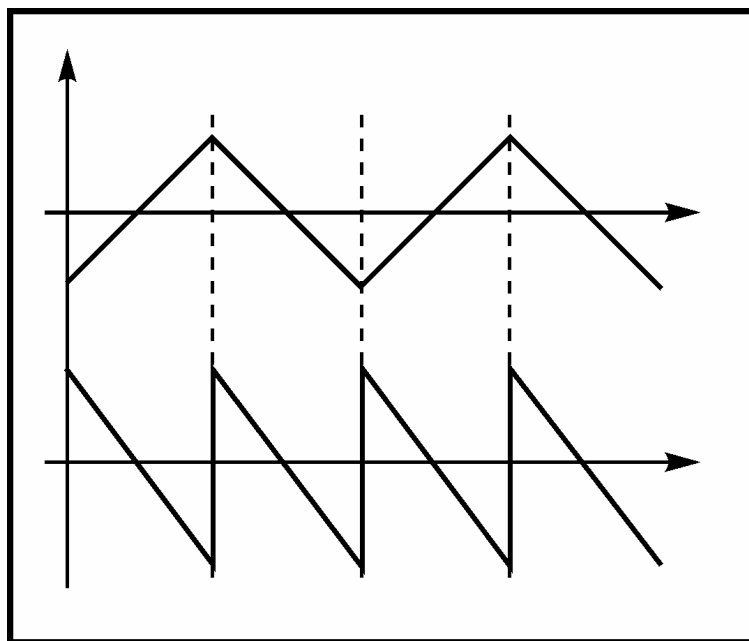
## 1.8 Ramp Wave Output

When I met the 566 some years ago now, I suppose I yawned. Perhaps you did too. At first blush it seems like a pretty pedestrian chip, sporting only the boring square and triangle wave outputs. And yet, as we've seen throughout this guide, it is possible to work some real magic with the 566 if you take time to explore, keep an open mind and try to ignore the dull examples given in the manufacturer's data sheets. The circuit of this section should really drive these points home.

As you probably know, it's easy to convert a ramp wave to a triangle wave. And in fact, many synthesizer VCOs take that route, using a ramp type generator at the heart and deriving all other outputs from it. But can one go the opposite direction, starting with a triangle and generating a ramp?

The answer in the general case would seem to be "fat chance." But thanks to the phase relationship of the square and triangle (at pins 3 and 4, respectively) it turns out to be remarkably easy with the 566. Refer back to *Figure 1.7-1*, and you'll notice that the square wave changes state whenever the triangle is at its highest peak or lowest valley. That's the secret.

What we'll do is force what would normally be the triangle wave at pin 4 to start over again from a full charge on the timing cap whenever it hits the lowest point. See *Figure 1.8-1* which illustrates this. It's as simple as forcing a transistor to dump a charge on the cap at the right moment. And that moment is indicated by the square wave on pin 3.



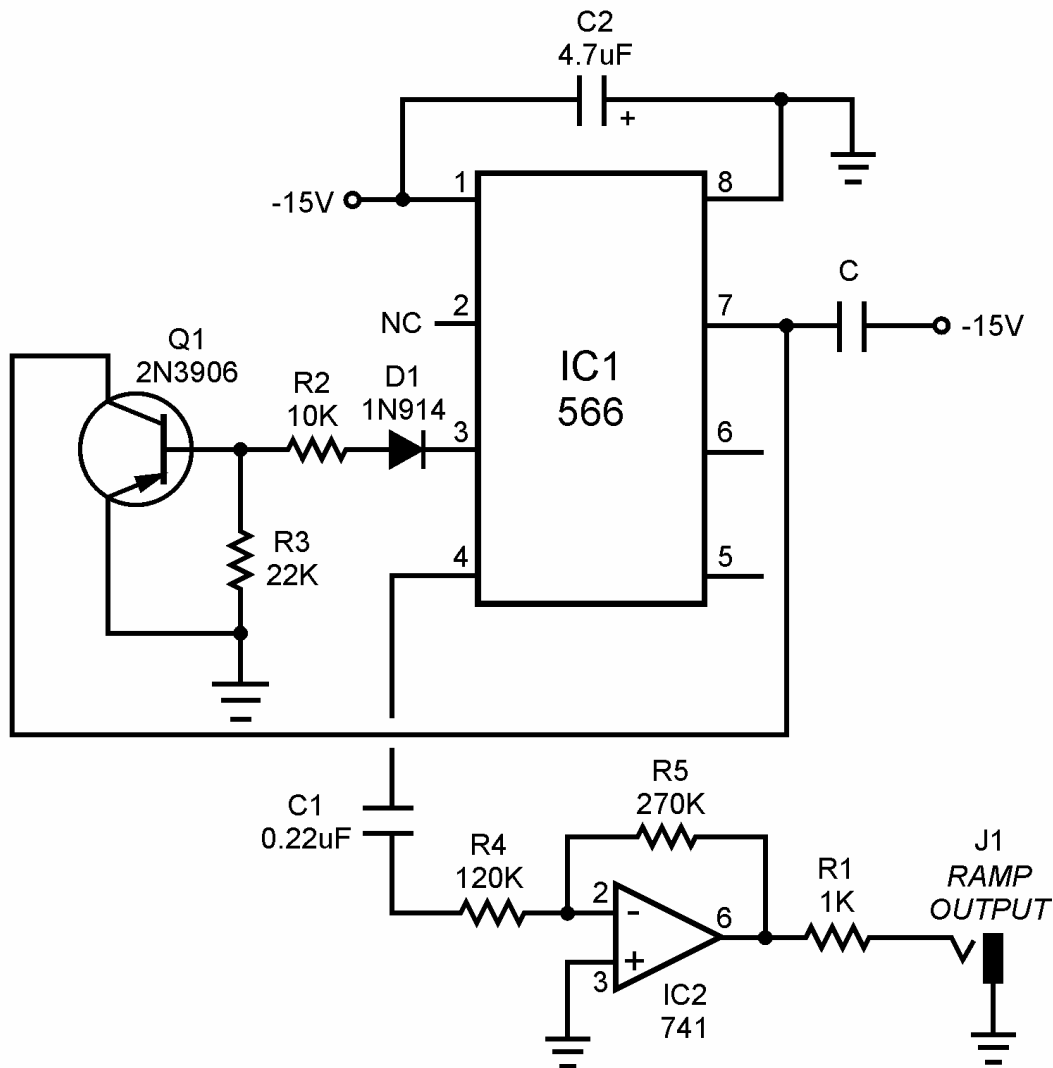
*Figure 1.8-1*  
*Triangle and Ramp Waves*

Look at *Figure 1.8-2* now. To keep things simple, the control circuitry isn't shown here, but you can apply this method to the designs in both Sections 1.x and 1.x. In either case, observe that we're employing a -15V power supply.

The square wave is monitored at pin 3. Diode D1 ensures that only negative pulses are passed on to transistor Q1. Voltage divider R2/R3 chops the signal down to a manageable level suitable for applying to the base. When a negative spike occurs, Q1 is turned on briefly. A large amount of current will flow from the collector and is dumped onto the timing capacitor C. In effect, the signal is forced to start from its highest point. The charge is slowly bled off via the usual current source inside the 566, and when it hits the lower threshold a new charge is dumped onto C and the process starts all over again.

In the ideal case the frequency of the ramp should be exactly double that of the triangle. However, owing to the behavior of Q1 and the fact that the additional charge is a bit larger than what the 566's current sources would normally deliver, the frequency is slightly less than double. Note too that the amplitude of the waveform at pin 4 is altered slightly. So, op-amp IC2 and its associated components scale things a bit to deliver the usual  $\pm 5V$  signal with a 1K impedance.





### Design Notes

- Note the unusual power supply connections to the 566.
- For simplicity, the control circuitry to pins 5 and 6 isn't shown. Refer back to Sections 1.2 and 1.3 for details.
- With the addition of Q1, which alters the charge cycle on C, the output frequency is approximately double what it would otherwise be.
- Q1 dumps a large charge on C, and this changes the amplitude of the signal appearing at pin 4 somewhat. IC2 and associated components scale this appropriately.
- The output is a nominal 10Vpp in amplitude and 1K in impedance.

*Figure 1.8-2  
Deriving a Ramp Wave*

## 1.9 Adding Hard Sync

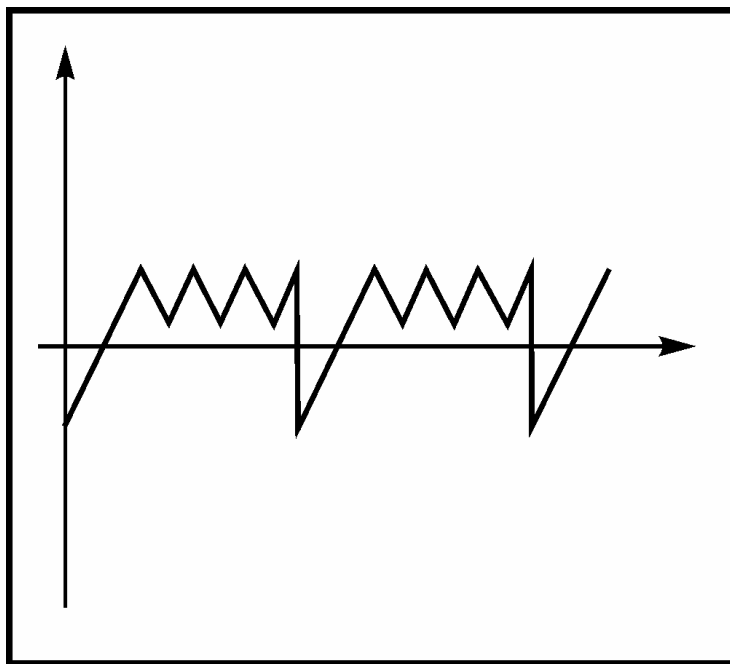
Talk about the "aha" experience! When I first saw (in an old manufacturer's data sheet) that it was possible to get a ramp wave out the 566 using a transistor switch, a flash of intuition hit me. It should be possible to coerce the chip to also sync to another VCO using a similar technique. In fact, that's the case and so this section is a natural extension of the last.

I think hard sync is one of the coolest musical effects around. I love the wonderful, throaty vowel sounds it creates.

The way in which it injects wild harmonics into an otherwise dull sounding triangle wave is amazing. *Figure 1.9-1* shows what it looks like on an oscilloscope. You can make out little ripples which are generated by the slave VCO. These represent, in effect, extra harmonics. The output of the master VCO, on the other hand, manifests in this diagram as the larger overall pattern; you can detect its two, longer ramp cycles in the basic shape.

Just to make sure that the concept is clear, the master VCO sets the fundamental frequency or pitch, and altering the tuning of the slave adds and subtracts harmonics to and from the fundamental.

Refer to *Figure 1.9-2* now, which shows the circuitry required to add hard sync to the 566. For simplicity, the frequency control business surrounding pins 5 and 6 isn't shown, but this circuit is compatible with the ideas covered in Sections 1.2 and 1.3. As usual, notice that we power the 566 from a -15V supply.

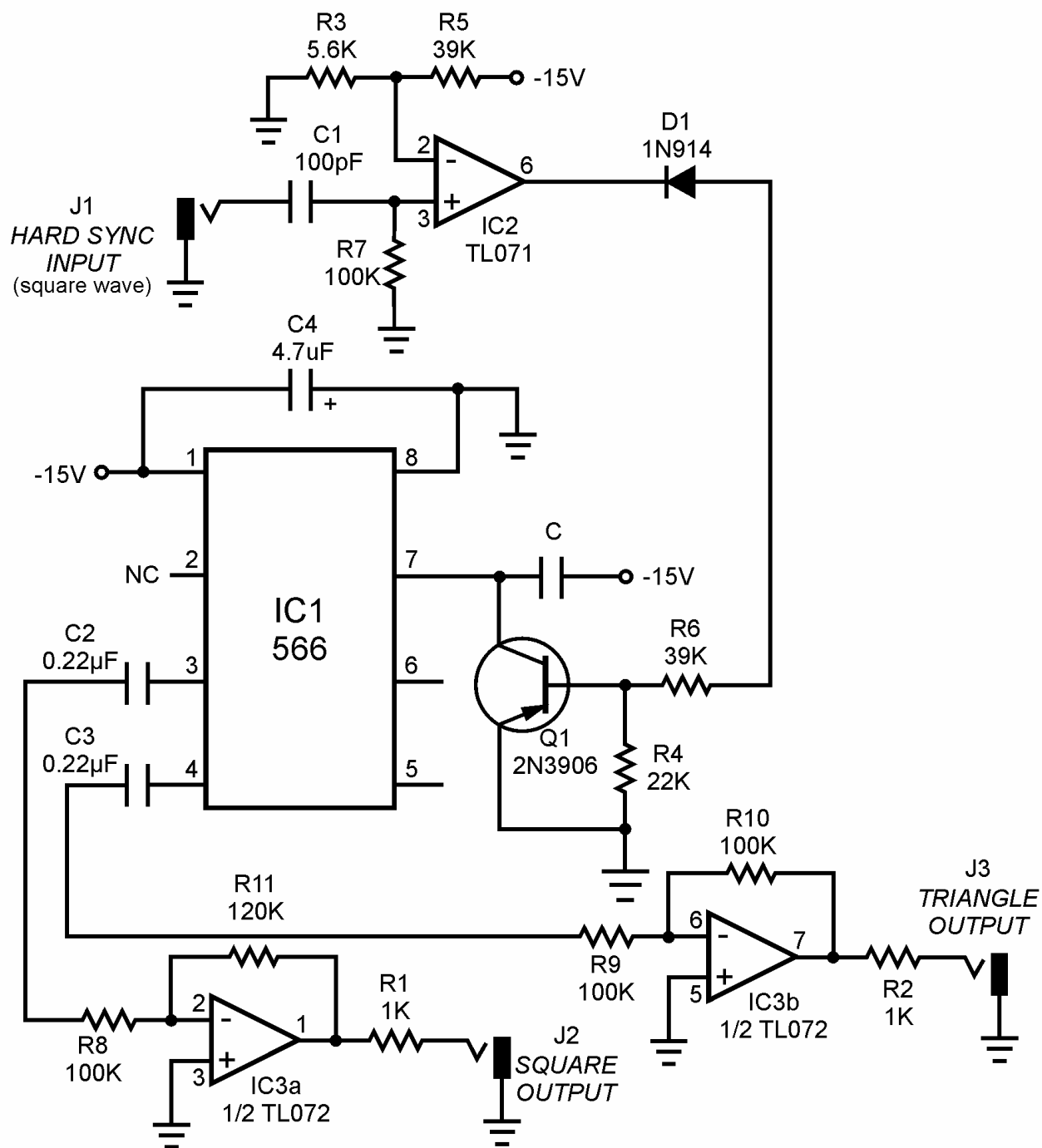


*Figure 1.9-1*  
*Hard Sync Waveform*

The square wave output of the master VCO is applied to jack J1. The differentiator comprising C1 and R7 converts this to an extremely narrow pulse or trigger signal. Voltage divider R3/R5 sets a threshold that ensures the pulse is negative going. Finally, D1 blocks the positive excursions altogether. The clock signal, then, is an exceedingly narrow pulse swinging from ground to about -15V. R4 and R6 chop this down to size, where it is then applied to the base of Q1. As in the ramp idea of Section 1.8, the collector current of Q1 is dumped onto timing cap C, forcing the 566 to reset unconditionally. This corresponds to the larger vertical strokes in *Figure 1.9-1*, above.

The additional voltage applied to C alters the amplitudes of both the square and triangle waves appearing at pins 3 and 4, respectively. So, op-amps IC3a and IC3b balance these out. Quite by chance, the modified triangle wave becomes a fairly good  $\pm 5V$ , so IC3b is simply set up as a unity gain amplifier. The square wave is a trifle weaker, so we give IC3a a gain of 1.2. The result is that the waveforms appearing at jacks J2 and J3 are the standard 10Vpp, with an impedance of 1K.

By the way, due to the nature in which hard sync works, the DC bias or baseline of either output is always changing. Hence, C2 and C3 are required to block any offsets.



### Design Notes

- Note the unusual power supply connections to the 566.
- For simplicity, the control circuitry to pins 5 and 6 isn't shown. Refer back to Sections 1.2 and 1.3 for details.
- The output is a nominal 10Vpp in amplitude and 1K in impedance.

Figure 1.9-2  
A Circuit to Add Hard Sync to the 566

## 1.10 A Tunable Noise Generator

A good white noise generator is an essential part of any analog synthesizer. This is especially true if you hope to come up percussive sounds. Now by definition white noise is a completely random mix of all frequencies, just as white light is a blend of all colors. On the other hand, pink noise is similar but more emphasis is given to the frequencies within the audio band. It's possible to carry this even further and create other types of noise by emphasizing or shunning various frequencies. This is usually done by following a white noise source with an active filter whose cutoff frequency can be adjusted. The filter can quickly become quite complicated and be touchy to adjust.

But here's a new approach to the problem. This section describes a tunable noise generator which instead of using an active filter employs FM (frequency modulation) to obtain a broad range of sounds. What comes out of it is noise, of course, but it's possible to emphasize certain bands of frequencies. When sweeping it over its range, the effect is quite ethereal - a sort of "swooshing" not unlike the sound of wind whistling through forest trees. Let's investigate the general procedure first before turning to the circuit itself.

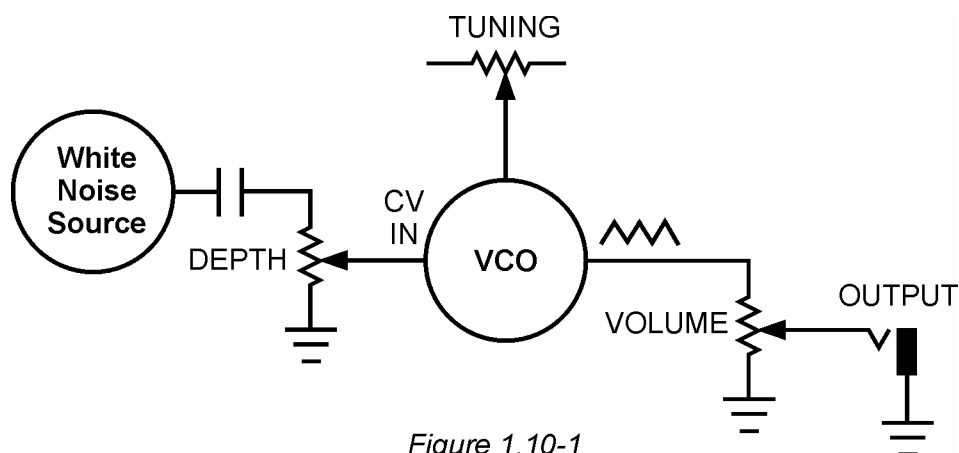


Figure 1.10-1  
Block Diagram of a Tunable Noise Generator

Figure 1.10-1 shows the block diagram of the tunable noise generator. First, white noise is created using standard techniques which then modulates the frequency of a VCO configured around a 566. The depth control is used to set the amount of modulation desired. A separate tuning control adjusts the center frequency of the VCO. The modulating white noise, then, causes the VCO to stray randomly from this center frequency. In this way a certain tonality is imparted to the sound thus created; it's still noise, but it seems to be focused around a certain band of frequencies. By twisting the tuning control it is possible to sweep the sound across the entire audio spectrum.

Incidentally a triangle wave seems to work best here, giving a fairly smooth sound. Square waves, on the other hand, lead to a rather harsh and gritty effect. In any case, the output of the VCO is dropped across the volume control which lets you set the amplitude.

So we have arrived at a unique method for "tuning" the response of a noise generator without requiring the use of active filters. Let's check out a practical circuit which implements the block diagram of Figure 1.10-1.

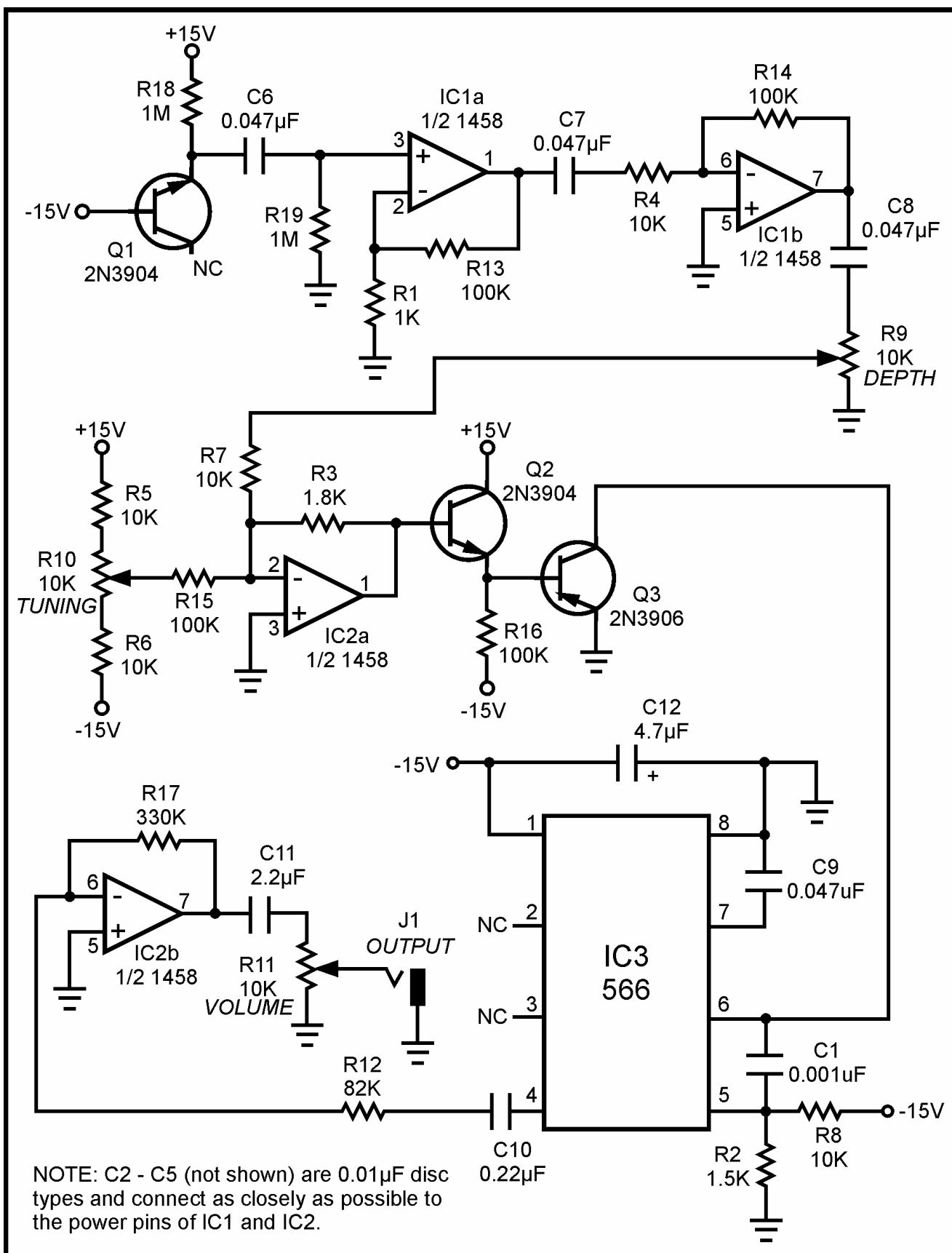


Figure 1.10-2  
Complete Circuit for a Tunable Noise Source

Refer to *Figure 1.10-2* which shows the schematic for the tunable noise generator. Let's start out by analyzing the white noise source first, leaving the VCO portion for later. Notice the unusual arrangement of transistor Q1. The collector isn't used at all, which suggests that Q1 is being pressed into service as a diode. In fact, it's really behaving as a Zener diode since the base-emitter junction has been deliberately reverse biased. Observe that the emitter has been tied high through R18, while the base connects to the negative side of things. Thus, the transistor is straddling a full 30V (+15V and -15V) which is enough to overcome the base-emitter breakdown voltage. This forces it into avalanche mode and produces a really decent source of white noise at the emitter. R18 limits the current flow through the transistor, so that it doesn't fry in the process.

R18 also acts as a load resistor for the noise thus generated. Unfortunately, the signal is a trifle weak just now and needs to be boosted. So, we send it to a preamplifier configured around IC1a, AC coupling it in the process by means of capacitor C6. Notice that C6 is purposely kept fairly small so that the bass frequencies will be attenuated a tad. This helps prevent rumbles in the sub-sonic region.

R1 and R13 set the gain of op-amp IC1a to a factor of 101. But believe it or not, that's still not enough "oomph" to properly modulate the VCO yet to come. So we move on to yet another preamplifier, this time comprising IC1b, R4 and R14. Operating in inverting mode now, this op-amp will have a gain of 10. Put the two preamps together, and we have boosted the white noise by a factor of over 1000. Now it's strong enough to do what we require of it!

The hefty noise available at pin 7 of IC1b may have accumulated an unwanted offset, so we AC couple it to the next stage via C8. The full strength signal is applied across potentiometer R9, which lets you manually adjust how deeply the noise will modulate the VCO.

The standard linear control voltage input at pin 5 of the 566 is too seedy for our purposes, so we'll use the exponential approach which was described in Section 1.3. Refer back there if you need to review how the exponential converter works.

R15 is driven by the wiper of R10, the tuning potentiometer. The top and bottom ranges of this control have been limited appropriately by R5 and R6, respectively. With this pot you can sweep the VCO's basic frequency from about 20Hz to 20KHz in one fell swoop; there's lots of usable range here!

But here is the key to the whole circuit. Recall that the white noise generator output can be picked off the wiper of R9, the depth control. This signal feeds R7, which then sums into IC2a. Thus the noise modulates the basic frequency of the VCO, completely at random. Potentiometer R9 lets you adjust how much modulation you want. Naturally, when set to zero the circuit performs as a normal VCO.

And that brings us up to the output of the entire circuit. There are two waveforms available from the 566 chip, a square wave (pin 3) and a triangle wave (pin 4). Feel free to use either waveform, but in general the triangle output sounds quite a bit smoother. But remember that pin 4 rides on a rather heavy DC bias which needs to be blocked, and the amplitude of the triangle there is a bit feeble. Capacitor C10 AC couples the signal to op-amp IC2b, which then boosts it up to about 10Vp-p. The output is applied to volume control R11, which can tame the signal as desired before it finally appears at output jack J1.

When using the tunable noise generator, be careful not to blowout either your loudspeakers or your ears! The output of this device is pretty hefty (up to a max of 10Vp-p). This higher value was selected so that the circuit would be compatible with analog synthesis equipment. On the other hand, if using the tunable noise generator with standard hi-fi gear, you'll want to turn volume control R11 down to give an output of around 2Vp-p.

# Bibliography

---

(Books and Articles Mentioning the 566)

- anonymous, *Organtua Assembly Drawings*, (Oklahoma City, Oklahoma: PAiA Electronics), 1978.
- anonymous, *Organtua Assembly and Operation Manual*, (Oklahoma City, Oklahoma: PAiA Electronics), 1978.
- Anderton, Craig, "Experimenting with the NE566," *Electronotes*, Volume 1, Number 8, June 30, 1972, p. 8.
- Henry, Thomas, *The Electronic Drum Cookbook*, (N. Mankato, Minnesota: Midwest Analog Products, 1996).
- Henry, Thomas, "Build a Tunable Noise Generator," *Nuts & Volts Magazine*, November 1999, pp. 25 - 27.
- Henry, Thomas, and Orman, Jack, "Build the ADV-Snare," *Nuts & Volts Magazine*, July 1996, pp.71-75.
- Henry, Thomas, and Orman, Jack, "Build the ADV-Bass," *Nuts & Volts Magazine*, November 1996, pp. 56 - 59.
- Jones, Marvin, "Build the PHLANGER for Dramatic Music Effects," *Radio-Electronics*, October 1977, pp. 42 - 45, 92, 93.
- Jones, Marvin, "Build This String Synthesizer," *Radio-Electronics*, February 1979, pp. 37 - 41, and March 1979, pp. 71 - 75, 104, 108, 110.
- National Semiconductor, "LM566/LM566C Voltage Controlled Oscillator," data sheet, *Linear Data Book*, (Santa Clara, California: National Semiconductor Corporation, 1982), pp. 947 through 9-49.
- Signetics Company, "NE/SE566 Product Specification," "AN185: Circuit Description of the NE566," "AN186: Waveform Generators with the 566," *Linear Data Manual, Volume 1, Communications*, (no place: North American Philips Corporation, 1988), pp. 4-290 through 4-298.
- Simonton, John, "Potpourri and the Apple Connection," *Polyphony*, November 1977, pp. 28 - 31.
- Wood, Steve, "Professional Drum Synthesizer," *Radio-Electronics*, May 1980, pp. 49 - 52, and June 1980, pp. 59 - 62.