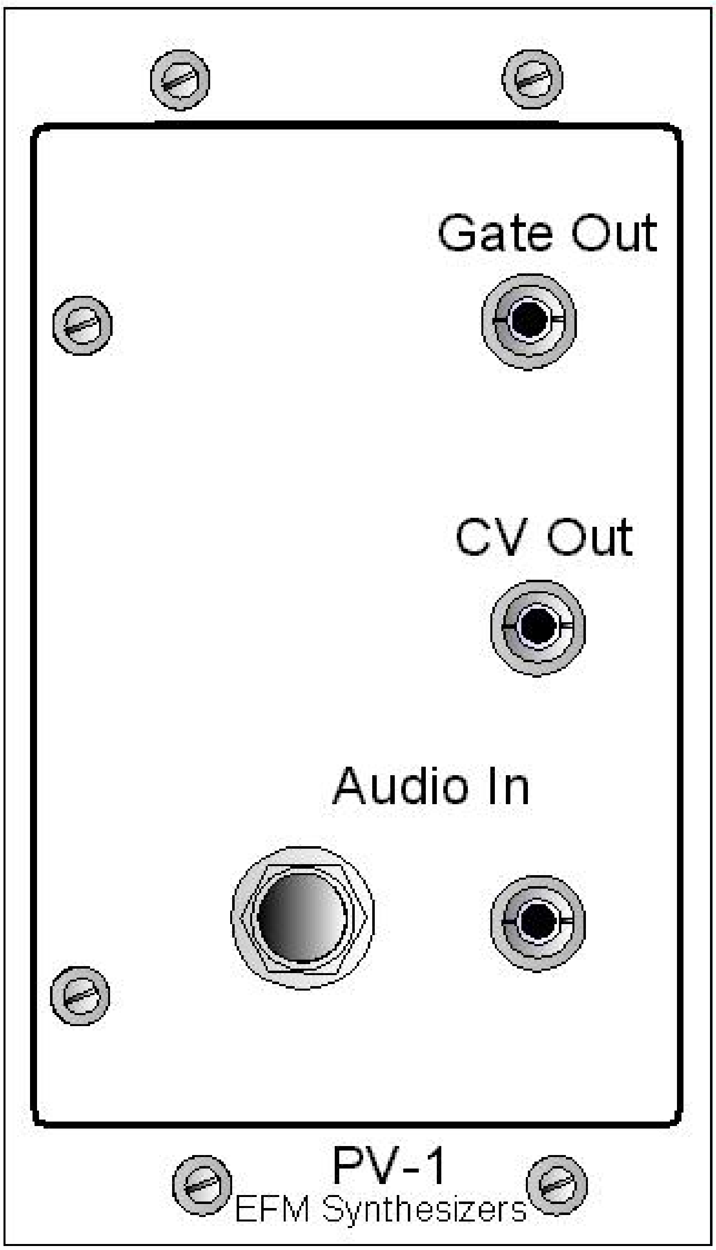
PV-1



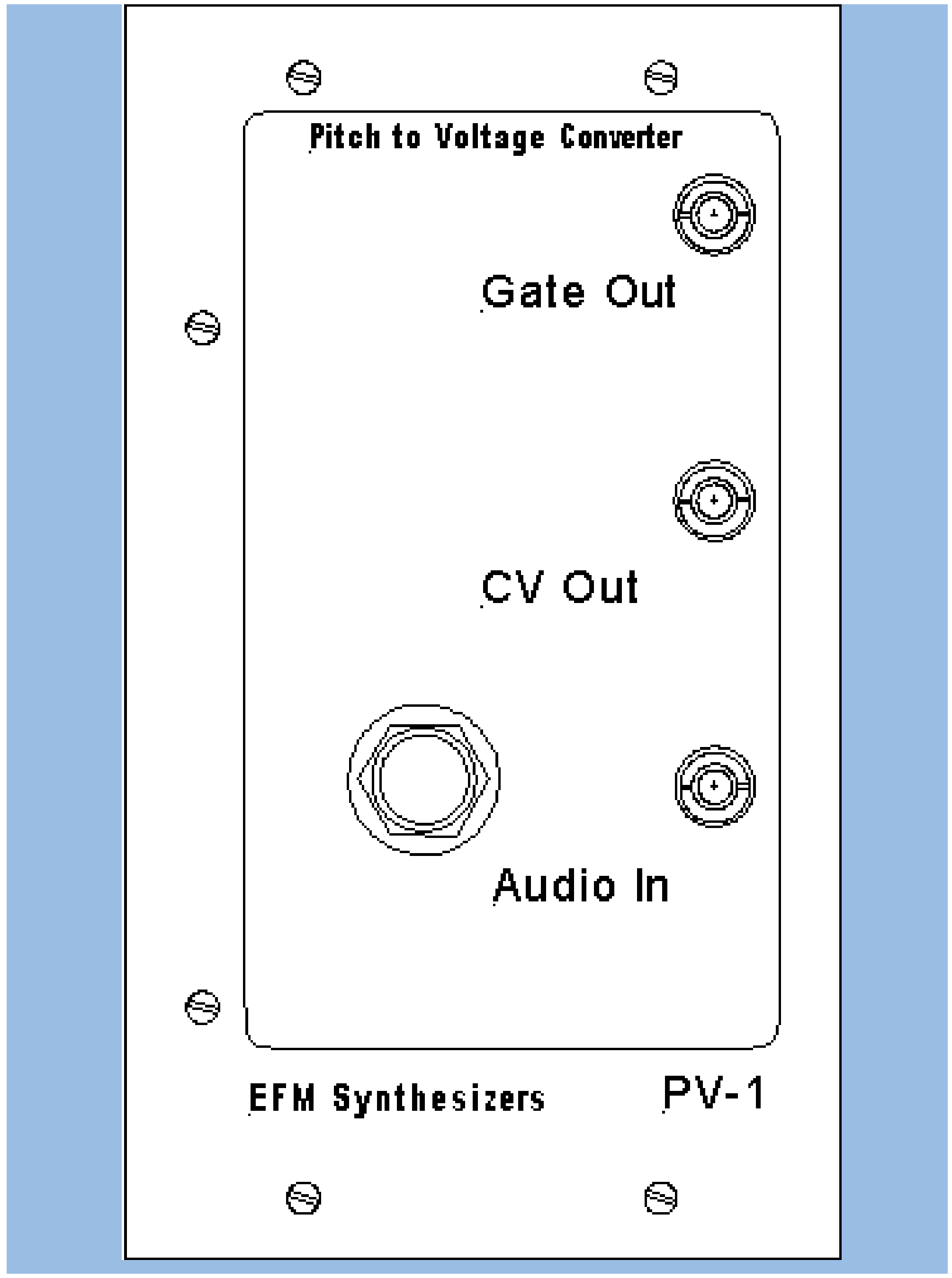
Pitch To Voltage Converter

Features...

- Excellent Tracking
- +/-12 or +/- 15V

PV-1 Pitch to Voltage Converter - By Harry Bissell V-1

Based on the "Etherwave Pitch to Voltage Converter" by Bob Moog/Big Briar Used with Permission:



C01 POLYESTER .22uF
C02,03,04 AXIAL CERAMIC .001uF
C05,06 POLYPROPYLENE .1uF
C07 CERAMIC DISK 100pF
C08 CERAMIC DISK 330pF
C09 AXIAL CERAMIC .001uF
C10 AXIAL CERAMIC .002uF
C11,12,13,14,15,16,17,18 AXIAL CERAMIC .1uF
C19 CERAMIC DISK .01uF
C20,21 ELECTROLYTIC 10uF 25VDC
D01,02,03,04,05,06 1N4148
P01 BOURNS 3296W-103 10K
P02 BOURNS 3296W-503 50K
PL01 4 pin .156 MOLEX or equivalent
PL02 8 pin .100 MOLEX or equivalent
Q01 2N3906
Q02,03,05 2N3904
Q04 (or similar) MAT02 (2SC1583)
R01,02 47K
R03,04,05,36 22K
R06,07,27,29,33 10K
R08,09 METAL FILM 681K
R10,R19,37 330
R11 150
R12,13,17 METAL FILM 22.1K
R14 METAL FILM 1.2M
R15,18 METAL FILM 49.9K
R16 PTC 1K *TEMPCO*
R20 METAL FILM 499K
R21,22 10M
R23 470K
R24,25,34 1M
R26 100K
R28 2.2K
R30 22K
R31 150
R32 1.1M
R35 1K
U01 LM13700 (or LM13600)
U02 4093
U03 4017
U04,06 TL062 (or TL082)
U05 4016
U07 4013

Unmarked resistors are 1/4W 5% Metal Film indicated here stability would improve Polypropylene or Polystyrene caps as shown

Trouble ??? e-mail harrybissell@prodigy.net

Introduction: What is a Pitch to Voltage Converter ?

A pitch to voltage converter takes a frequency input and converts it to a voltage output. There are several forms of this circuit, both digital and analog. Most
of these can be grouped into two basic classes. The Tachometer circuit, and the Ramp Sample/Hold.

The Tachometer circuit is the simplest. The frequency input is converted to pulses of a fixed width, and these pulses are integrated, or averaged over time. There is a trade off between how fast you can get to a new level, and how much feed throughof the input frequency you can accept. The less ripple you wantin the output, the slower the circuit responds to a change ininput frequency. Tachometer circuits are best for frequencies that :

1) You want to average over a long time anyway

2) Frequencies that are continuous (do not start and stop...)

Note that things like motors etc... are the most common use for Tachometer circuits.

The best example of the Tachometer circuit as applied to Electronic Music is the Korg MS-20 external signal processor. It features a filter at the input and output that are linked... so the user can set the tradeoff between ripple and acquisition time to best fit the signal they are processing. The circuit also has an extended range Tachometer circuit... at very low frequencies the pulse width is fixed. As frequency increases, these pulses get closer together. At even higher frequencies the pulses overlap, and occasional gaps appear in between multiple pulses. This allows
the circuit to perform at frequencies above the normal range of the Tachometer circuit. There is still a very real delay in how fast the circuit can track a rapidly moving input frequency.

The Ramp - Sample/Hold is another class of circuits used to solve the problem of slow signal acquisition. The input frequency is divided in half, and then a linear ramp is generated for one cycle, and then sampled and held during the following cycle, and the ramp is reset. This happens over and over again... so the output is always one cycle behind the input. No matter how large the step in input frequency (within reason) the output will follow within two cycles of the fundamental.

The advantage of the Ramp circuit is speed of acquisition, and lack of ripple. The drawback is that it is more complex, and sensitive to noise in the input frequency. This approach was used in the 360 Systems "Slavedriver" Guitar to CV interface (mid 1970's) with some success. The noise problems prevented the circuit from achieving satisfactory performance at that time. Speed of response is still an issue.

Consider the lowest note of a Guitar = 80Hz, which has a period of 12mS. The Ramp circuit needs at least twice that time to process the Pitch Voltage, or 24mS. This is a physical limit for converting the pitch. If you have a lot of noise in the signal (harmonics etc.) it will take even longer.

The Pitch to Voltage converter must have an accurate reference to derive the pitch from. Harmonics will cause false operation, so inputs from real instruments like guitar, voice, etc. usually need signal processing before the pitch conversion.

About this circuit:

The PV-1 is a general purpose Pitch to Voltage converter based on Bob Moog's design for an accessory to the Etherwave Theremin, manufactured by his company Big Briar. I've used this as the basis for the PV-1, and in fact I've included everything necessary to hook up directly to the Etherwave. There are some circuit additions that will allow other uses. The circuit can be used to slave a Volt/Octave VCO to another source, like a Volt/Hertz VCO. This might be useful in slaving a more common synths to the Korg instruments. In the case of the Theremin or synthesizer VCOs, there is no need for extra filtering because they are continuous waveforms that do not vary in shape or amplitude.

There are also some "hooks" I've included for those who want to try their luck at tracking guitars etc... but that is an advanced project and not easily done. The idea was to have the basic P/V core as a simple printed circuit board.

How it Works:

The frequency input is applied at PAD1, through a high pass filter C01, R01. This removes any DC component of the input frequency. U01a is a limiting preamplifier. The OTA is overdriven resulting in a smooth, somewhat "squarish" wave of about 20V pk-pk. R04 and R05 form a voltage divider to limit the input to U02 NAND schmitt trigger (pins 1,2) to positive voltages only. U02 squares the input wave, and it is fed to U02 (pins 5,6) which invert this square wave. There are two RC networks (R06,C03 and R07,C04) which differentiate the negative going edges of these square waves. These negative going pulses
are applied to U02 (pins 12,13) which outputs a positive pulse at each zero crossing of the input frequency.

The zero crossing pulses are used to increment counter U03. This is a "Johnson counter with decoded outputs". The important point of this counter is that the output advances with each input pulse and there is no overlap between the output pulses. The counter starts at 0, and counts to 3. When the fourth output goes high, the counter is immediately reset back to zero. I've added an "or-gate" in this reset that I'll discuss later.

The counter controls the four states of the machine. During the first time period, ramp capacitor C05 is held in the discharged state by analog switch U05 (pins 1,2,3,4). At this time the sample and hold capacitor C06 is holding the last voltage that was applied.

During the next two time periods, the emitter of Q01 is driven positive. This sources current to the Gm input of U01b. Resistive divider R12, R13 holds the positive input of U01b at 1/2 the supply voltage. While the Gm input is biased on, U01b forms a constant current source that charges capacitor C05 in the positive direction.

During the fourth time period, the emitter of Q01 is held at ground, disabling the Gm bias to U01b and ending the charging of C05. The voltage level at C05 is now held at whatever level it was charged to during the first two periods. C05 voltage is directly proportional to the period of the input waveform (the longer the period, the higher the voltage...). This voltage is buffered by opamp voltage follower U04a. The third output of the counter is now high, which closes analog switches U05 (pins 8,9,10,11) and allows Sample/Hold capacitor C06 to charge to the level held on C05 (Sample mode). R11,R31, and D01 limit the peak current to the analog switch, and clamp the input to the analog switch to ground if the output of U04a should go negative (during power-on). U04b is an opamp voltage follower buffering the voltage on C06.

If the input frequency is still present, this cycle repeats with the last voltage held on C06, and capacitor C05 reset to zero volts.

The purpose of the diode "or-gate" that has been added is toallow some external circuit to reset the counter to the initialstate. This is not important in the case of the Theremin, or if your input is from a VCO. These sources run continuously (or nearly so). It is important if you intend to use external signal processing, as in the case of the guitar. In this case, the circuit must always start with the ramp reset. The next two clock pulses are measured for the period of the waveform, and then the sample is presented to the output. Should a new note occur at any time, the user can detect this "note on" condition and reset the pitch to voltage converter, while holding the last value in the sample/hold. This will minimize the glitches that might otherwise occur if the counter is allowed to remain in a random state.

The output of the Sample/Hold is presented to logarithmic converter. Pitches that are from "musical" sources have an exponential relationship to each other... each octave is one-half the period of the lower octave. To linearize the voltage output from the Sample/Hold, a log function is needed. The best description of how these converters work can be found in National Semiconductor App Notes 311 - Theory and operation of Logarithmic Converters. The Log converter uses a dual transistor such as the MAT-02, MAT-01 from Analog Devices, or the LM394 from National Semiconductor. A temperature compensating (tempco) resistor (R16) is required. These are available from various sources including Bob Moog's company, Big Briar.

R16 (tempco) is designed so that it can be epoxied to the top of Q04, for good thermal contact.

It would also be a good idea to pot this transistor and tempco together... but some experimenters frown on this because the components are so high priced and hard to find. It will work much better. Thermal mass in this area means that the transistor must change temperature slowly, and makes it more likely that the tempco resistor will track. Even isolating these components with some Styrofoam over them to prevent air drafts will help a lot.

If the output voltage from U04b (pin 7) is inverted it should be possible to directly drive a linear (V/Hz) oscillator. I haven't tried this.

Gate Circuit: The gate circuit used on this board is different from the Bob Moog design. I changed it to allow easier use ingeneral purpose applications. During the "0" time slot the counter is holding the Ramp in the reset state, and the Sample/Hold is holding the last value, if any. When the counter steps to time slot "1" D Flip-flop U07a is clocked. This causes the Q output (pin 1) to go high, charging C10 to Vdd (12 volts) quickly through D04, and charging C09 slowly through R30. This state remains until the voltage at C09 reaches the threshold voltage (1/2 Vdd = about 6 volts) at the reset pin of U07 (pin 4). The Q output then goes low, discharging C09 rapidly through D03, and discharging C10 slowly through R21.

U07b is clocked when the counter enters state "3" At this point the data is valid at the sample/ hold stage. If the voltage on C10 is still greater than the threshold voltage at U07b (pin 9) at this time, then the U07b Q output (pin 13) goes high, indicating that a valid frequency is present at the audio input. If the frequency is too low... C10 discharges below the threshold point and the Q output stays low. The time constants have been chosen to make this detector output a "1" (high level) when the input frequency is greater than about 20 Hz. Making C10 smaller will raise this frequency if desired.

The other detector consists of Q02 and Q03. This detector outputs a "1" (high level) whenever the amplitude control voltage is greater than about 1/2 volt positive. When both detectors are "1" (high state) NAND gate U02c (pin 10) goes low, removing the base drive from Q05.

Construction Notes:

This is not a beginner project, although it is straightforward. All of the components needed to use this circuit with the Etherwave Theremin are included on the board. There is an 8 pin .100" header on the board for connection to the Etherwave. The "Volume CV Output modification" for the Etherwave is included. This section has the audio output (pre VCA) available on Pad6 (AUDIO). You have to jumper this to Pad1 (AUDIO IN). Pad5 (VOL CV OUT) must be jumpered to Pad2 (VOL CV IN). The PITCH CV and GATE OUT should be wired to external jacks. You probably want to wire VOL CV OUT as well.

For Etherwave use the circuit can be mounted in a box beneath the existing case. I'm using a Hammond Mfg 1590D-BK which is a 7.4" x 4.7" x 2.1" cast aluminum enclosure with a black finish. Eagle makes a similar box that is black nylon coated... Bud has the same unpainted.

Aluminum is hard to paint so I went for this box. I removed the microphone stand adapter from the Etherwave, and screwed the box to the bottom with the circuit facing down... Mount the stand adapter to the box cover and you can easily get at the circuit boards for mods... calibration.. whatever...

For modular use you will probably not need most of the components in the "gate" circuit.

Just remember to not leave any CMOS inputs floating if you delete parts... and put in all the jumpers so that you don't accidentally miss any ground or power connections. The reset input is active high, a positive input will put the hold the last voltage, and reset the ramp cap. The first pulse that arrives after reset is removed will start the ramp timing cycle.

The unit can be used to process external instruments. They must be filtered to recover the fundamental, and they must either have a very fast decay time (less than 2 cycles of the highest frequency) or you must use the reset circuit to hold the last valid sample.

Those who are interested in using the PV-1 to process external signals (especially guitar) may contact me at harrybissell@prodigy.net. This will be a "very" advanced
project, not suited for those who do not have considerable skill with large projects.

Calibration:

The PV-1 has two adjustments Range (or zero) and Scale. There are many ways to calibrate depending on what instruments you have. You will need

1) a stable source of frequency... a VCO or Signal Generator may be used
2) a way of measuring that frequency, such as a scope, frequency counter or (my favorite) a chromatic tuner
3) an accurate digital voltmeter.

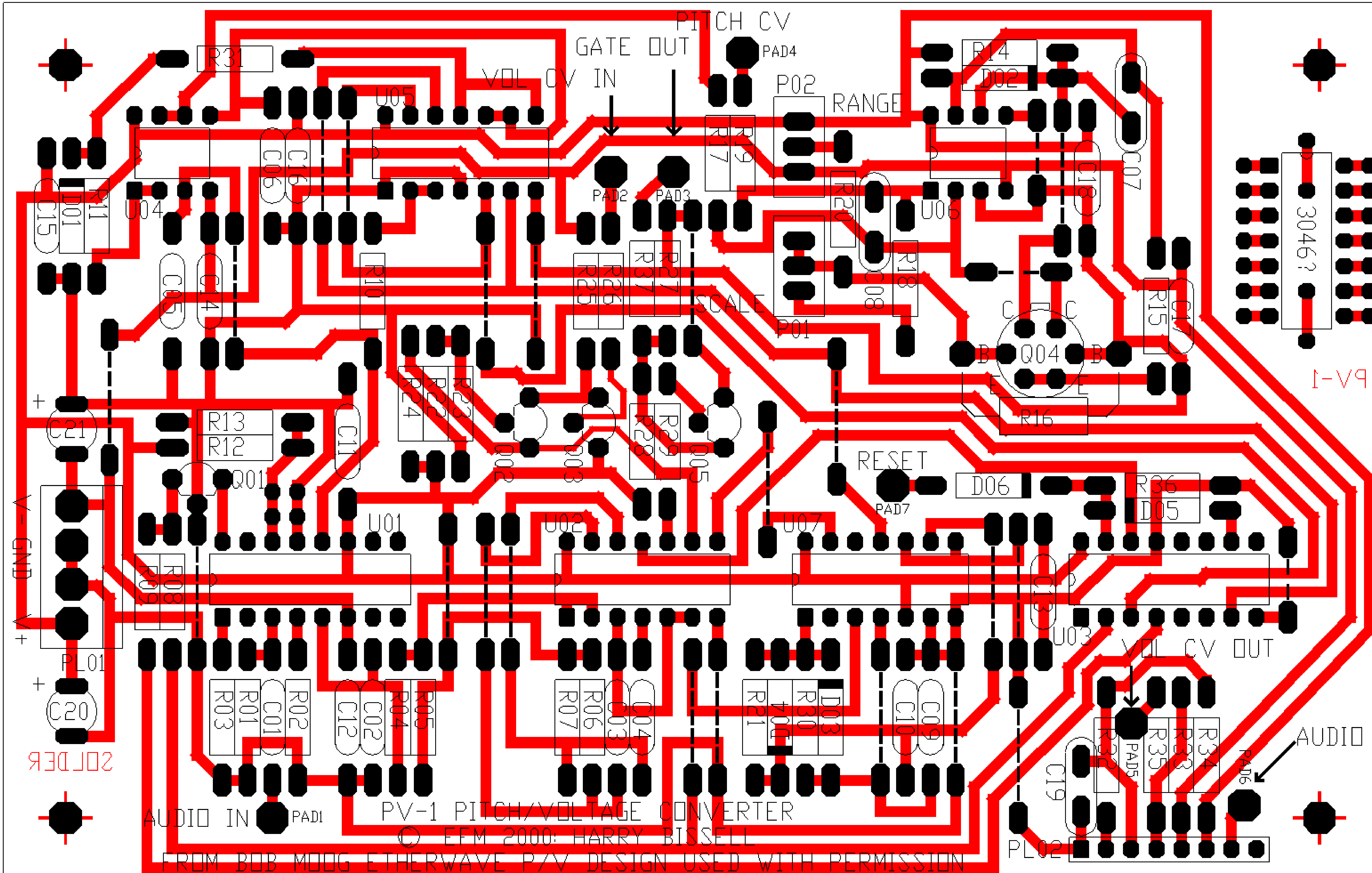
Method:

1) Set the frequency source to 130.8 Hz. (C BELOW middle C) Measure the Pitch CV Output. Adjust the RANGE trimmer until the meter reads 0.00 volts DC.
2) Set the frequency source to 2080 Hz (C two octaves ABOVE middle C) Measure the Pitch CV Output. Adjust the SCALE trimmer until the meter reads 4.00 volts DC.
3) Repeat steps 1 and 2 above. The adjustments interact, so repeat until no further improvement can be made. This will take several passes through steps 1 and 2.

Note: If you can't get the calibration to work, reset the trimmers to the middle and start over. Its possible to get yourself so out of range that calibration can become difficult.

Trouble ??? e-mail harrybissell@prodigy.net


PV_1 PITCH TO VOLTAGE CONVERTER
COPYRIGHT EFM 2000: HARRY BISSELL
BASED ON DESIGN BY BOB MOOG / BIG BRIAR
ETHERWAVE PITCH TO VOLTAGE CONVERTER
USED WITH PERMISSON
ETHERWAVE HEADER
PL02
PAD6
R34 1M
R33 10K
R35 1K
PAD5
R32 1.1M
C19 .01uF
PREAMP / LIMITER
LM13700
U01
R02 47K
PAD1
C01 .22uF
FROM AUDIO INPUT
R01 47K
R03 22K
R05 22K
R04 22K
C02 .001uF
4013
U07
R21 10M
D04 1N4148
C10 .002uF
D03 1N4148
R30 22K
C09 .001uF
R06 10K
R07 10K
4093
U02
C03 .001uF
C04 .001uF
ZERO CROSS
4017
U03
CK
CKEN
RST
CO
D05 1N4148
R36 22K
D06 1N4148
PAD7
RESET
U05
4016
FREQ > 50Hz
CV > 0.5V
R29 10K
R28 2.2K
Q05 2N3904
GATE OUT 0-10V
R37 330
PAD3
R27 10K
R24 1M
R22 10M
R23 470K
Q02 2N3904
Q03 2N3904
R26 100K
PAD2
FROM VOLUME CV
R25 1M
R12 22.1K
R13 22.1K
CURRENT SOURCE
LM13700
Gm
R08 681K
R09 681K
Q01 2N3906
RAMP
C05 .1uF
POLYPROPYLENE
R10 330
U04
TL062
R11 150
D01 1N4148
R31 150
S/H
C06 .1uF
POLYPROPYLENE
ANTILOG GENERATOR
D02 1N4148
C07 100pF
C08 330pF
R17 221K
R14 1.2M
Q04 MAT01
U06
TL062
R15 49.9K
R16 PTC 1K
PITCH CV OUT
PAD4
R19 330
R18 49.9K
P01 10K
R20 499K
P02 50K
PL01
C11 .1uF
C13 .1uF
C15 .1uF
C16 .1uF
C18 .1uF
C20 10uF
C12 .1uF
C14 .1uF
C17 .1uF
C21 10uF
U02
U05
U07
U03


PITCH CV
GATE OUT
VOL CV IN
RANGE
SCALE
RESET
AUDIO IN
AUDIO
VOL CV OUT
PV-1 PITCH/VOLTAGE CONVERTER
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