Stompboxology™

Volume 13, Number 3

Frequency Division & Subharmonic Synthesis

Analog frequency dividers generate a tone one or two octaves below an input tone. The sound depends on the shape of the new tone, and upon how well it tracks input dynamics. Squarewaves are easy to generate but sound harsh and don't track. More complex dividers yield smoother tones with excellent tracking. Dividers crop up in stompboxes, and in rackmount tools called subharmonic synthesizers.

This issue explores frequency division and subharmonic synthesis, and presents plans for both effects.

Frequency Division Techniques

The simplest way to halve a pure tone is to convert it to a squarewave, then divide the squarewave by two. Halving again makes a subharmonic two octaves below the input. Circuits that use these squarewaves as musical signals are called *direct dividers*. Sine-to-square converters (SSCs) and divide-by-two circuits are well known in electronics; Fig. 2 gives examples of SSCs; Fig. 1 shows a common ÷2/÷4 block.

Indirect dividers also derive squarewaves at half the input frequency, but employ the squarewaves as control feeds. One approach combines the halved squarewave with the primary tone in a circuit known as a *synchronous detector*. The output contains terms representing the sum (x + 1/2x = 3/2x) and difference (x − 1/2x = 1/2x) of the primary tone and its halved squarewave. The difference term is the subharmonic. Waveforms convey an intuitive understanding of this process (Figs. 4–1 & 8). Compared to direct division, the output contains much less energy as high/odd harmonics, but still demands lowpass filtering to isolate the subharmonic.

Another indirect divider slices the input waveform at a rate half its own, in a circuit that halves gain and applies a DC offset. The resulting segments combine to mimic a sinewave of half the input frequency (Figs. 4–2 & 9). Of the methods under discussion, this one's raw output most resembles a sinewave; post-filtering is still required. The technique is found in a pedal of current manufacture (Ref. 7). Both indirect dividers preserve dynamics of the input tone.

Problems and Limitations

The three dividers just discussed process a single tone at a time. This poses a problem for stringed instruments, whose tones consist of a fundamental plus *harmonics*, which are integer multiples of the fundamental. Fundamental and harmonics swap dominance as the note decays. SSCs skip octaves because they key on amplitude. Lowpass filtering ahead of SSC reduces the tendency to skip, but no single filter strips harmonics while leaving fundamentals. And most guitars contain open strings or dead frets where the fundamental is vestigial at best. Avoidance of octave skipping involves several steps: (1) lowpass filtering the input; the optimum filter varies because guitars differ in harmonic profile and dead frets; (2) incorporating hysteresis into the SSC; (3) tweaking the axe's tone pot sometimes helps; (4) finally, altered playing strategy is usually necessary; the player first explores the fretboard to identify dead strings and frets, then moves up the neck to play the weak notes on lower strings.

In this issue:

A second problem occurs as a decaying note nears the SSC threshold. Instead of fading out like a normal note, the squarewave train sputters and dies. Solutions include downward expansion or gating; or using an indirect divider, which is less prone to sputter audibly.

Third, direct-divider boxes sound unnatural due to squarewaves' constant amplitude. A more natural sound results from dynamic expansion of the squarewave, which also reduces audible sputtering as the note decays.

Last, direct-divider effects sound harsh. This suits them to distortion, especially for generating a gravelly sound. Smoother sound calls for lowpass filtering, an imperfect solution often taken in trade for simplicity. While a single sinewave lives at the squarewave fundamental, isolating the sine takes a fairly narrow bandpass. Lowpass filtering yields a spectrum of tones whose shape and amplitude vary with frequency, but which in practice never completely lose the buzz. (An exception occurs in subharmonic synthesizers, where the tones of interest fall in such a tight band that one filter converts squarewaves to sinewaves.) More effective—but complex—approaches involve tracking filters that automatically tune themselves to the input fundamental.

Subharmonic Synthesis

Subharmonic synthesizers and stompbox dividers have different goals. Where divider boxes create tones that don't exist, subharmonic synthesizers, at least early on, were said to restore bass rolled off to accommodate analog recording media. The reasoning held that tones of 40–100 Hz represented second harmonics of bass fundamentals poorly supported in vinyl records and analog tape. Users found subharmonic synthesis an effect desirable in its own right, whether the new tones were restored fundamentals or not. Subharmonic synthesis found a niche reinforcing the beat in dance clubs and augmenting motion picture sound effects.

Consumers got their hands on this tool late in the seventies when dbx® introduced the first of several subharmonic synthesizers, one of which remains in production. The original circuit used six bandpasses about 10 Hz wide (Ref. 1 & Fig. 3). Each bandpass fed a block containing a SSC, a ÷2 circuit, and a synchronous detector. The six summed subharmonic outputs were filtered to remove terms above 55 Hz, then mixed with the origi-

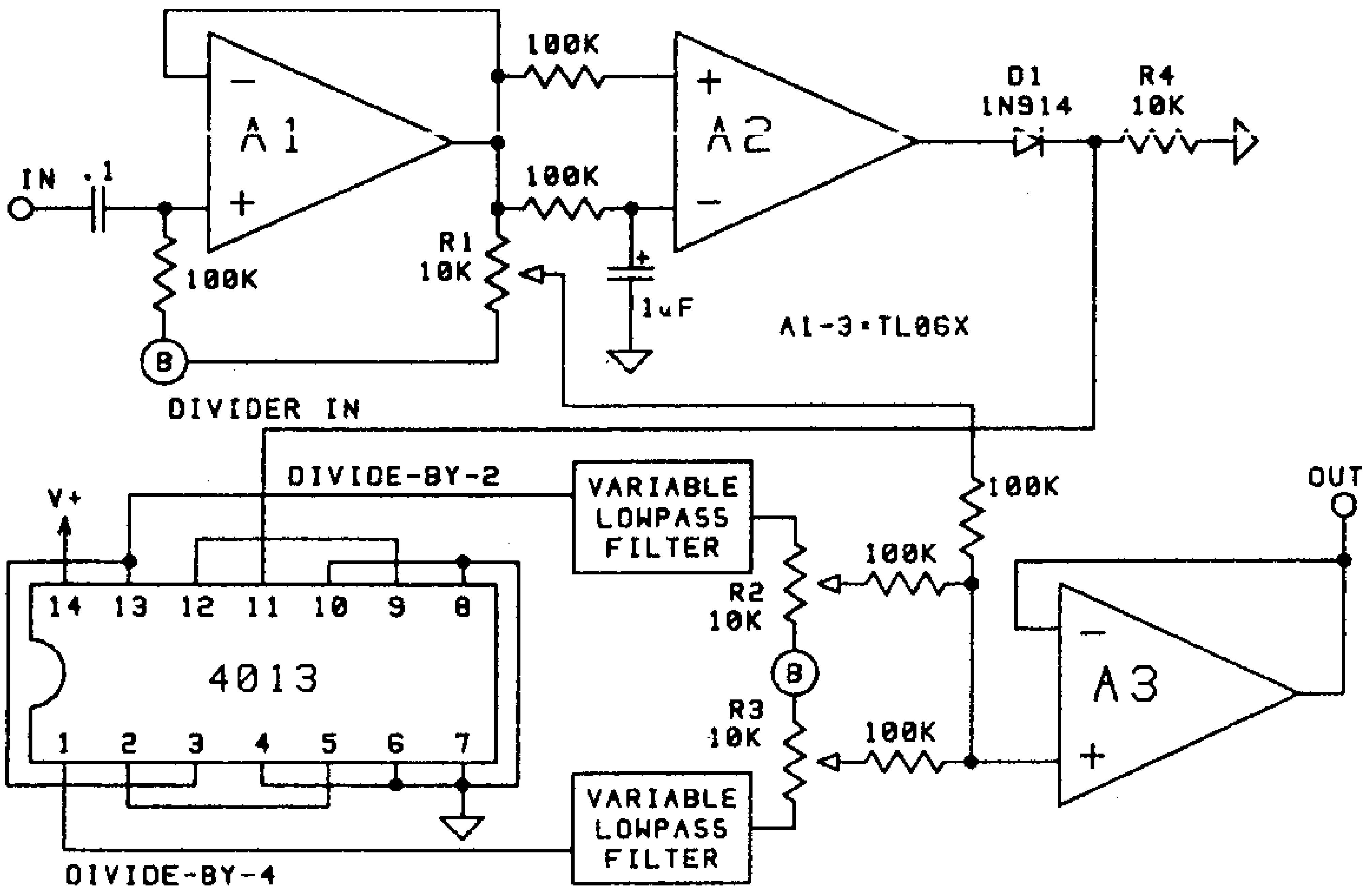
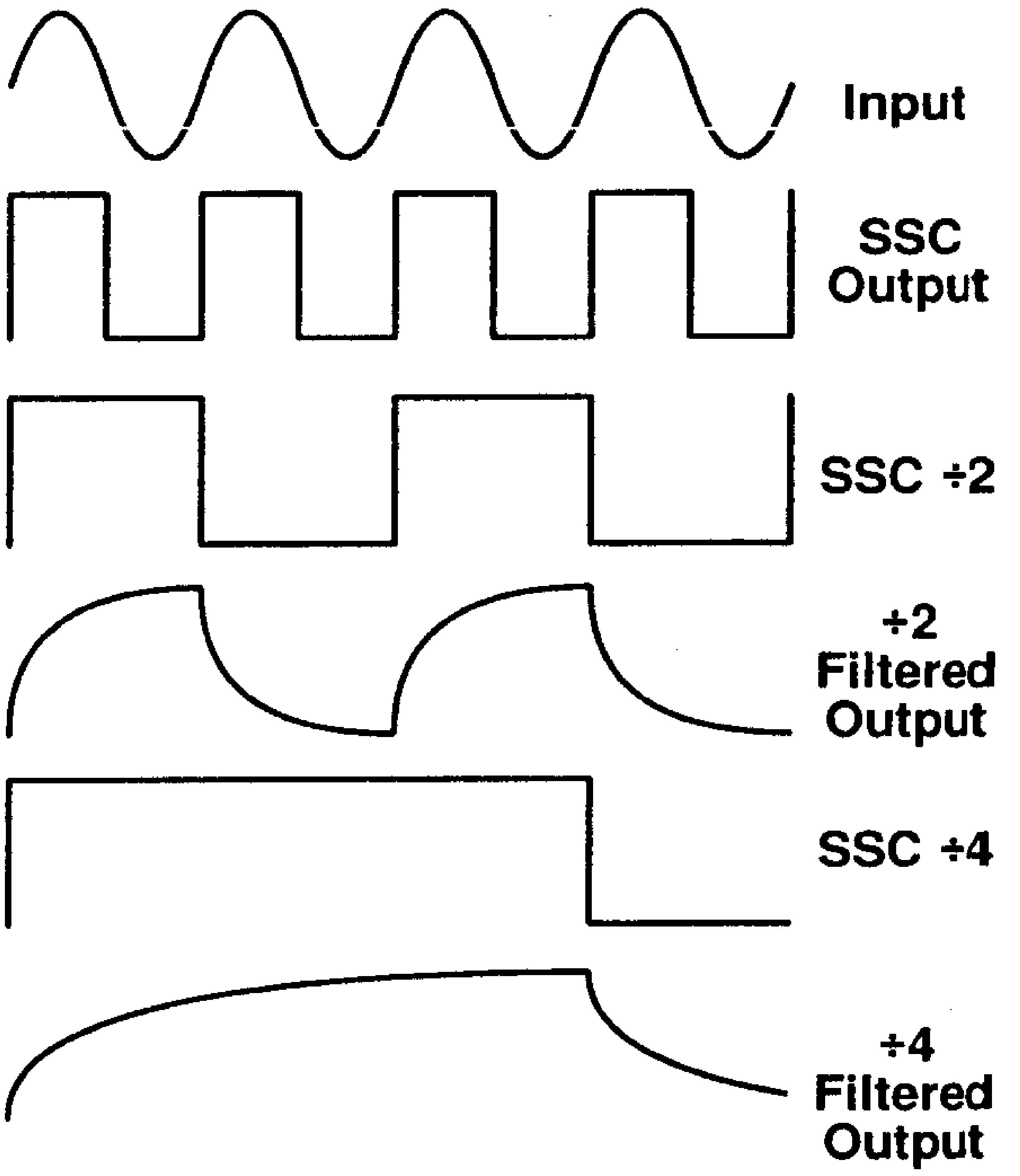


Fig. 1. Divider demonstration circuit. Sine input is buffered by A1, drives sine-to-square converter A2, whose squarewave output feeds 4013 dual flip-flop configured as a pair of divide-by-2 circuits; output of first divider feeds second, yielding divide-by-4. Each output negotiates a variable first-order lowpass filter; R2 & R3 control relative levels of ÷2 and ÷4 signal in output mix. This circuit is not suitable for use with guitar, because unfiltered tones result in severe octave skipping.



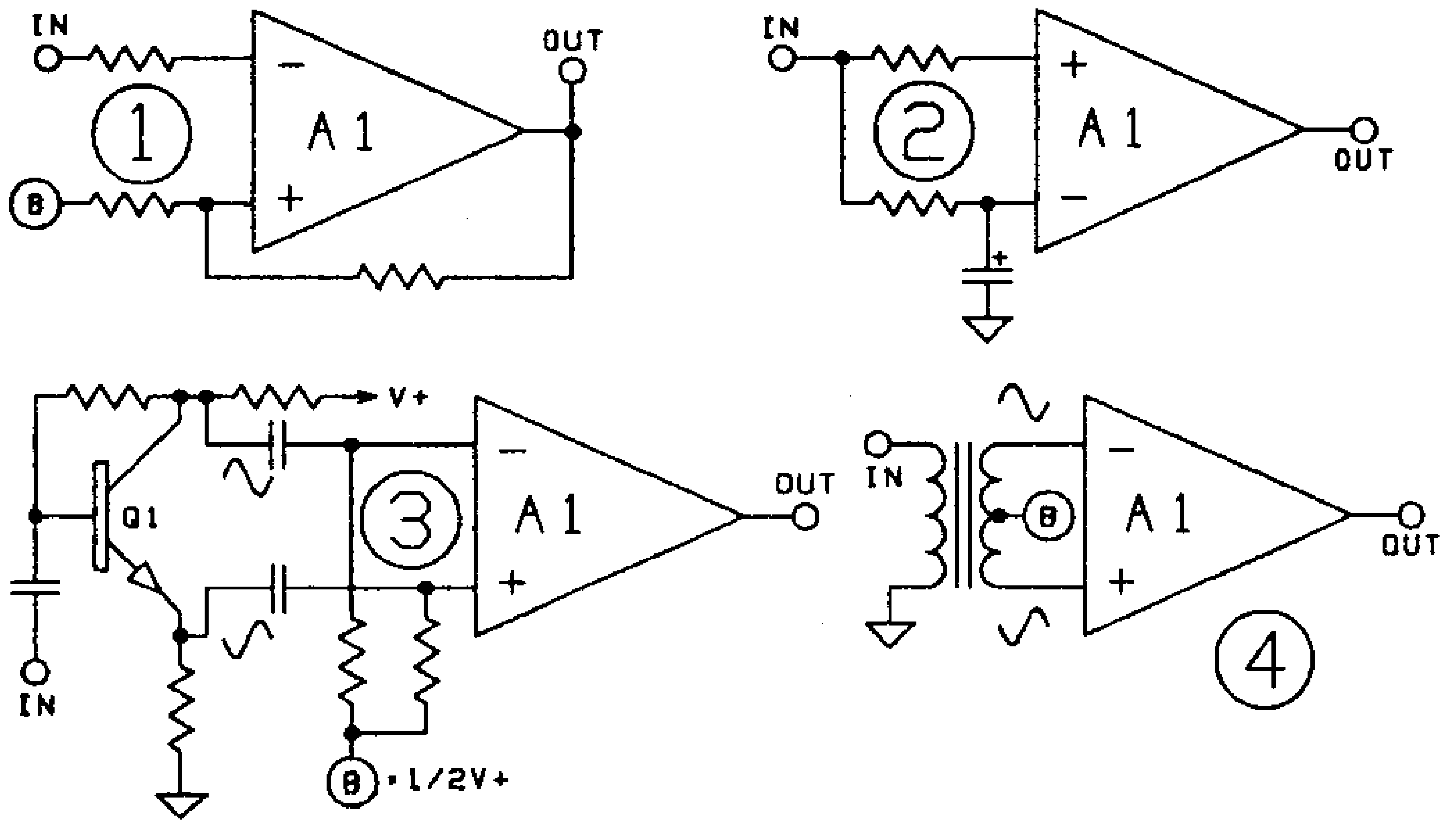


Fig. 2. Common sine-to-square converters. 1—Comparator configuration sometimes called Schmitt trigger. 2—Another comparator; lag network formed by resistor and capacitor causes changing signal at inverting lead always to trail that at the noninverting lead. 3 & 4—Comparators preceded by phase splitter. Given proper values of coupling components, these comparators can act as zero-crossing detectors.

nal signal.

An alternative approach emerged in 1987 (Ref. 2). Rather than a plurality of bandpasses, each creating a subharmonic, this method relied on a single bass bandpass (Fig. 5) feeding a synchronous detector. The reasoning held that most recorded music contains only one dominant bass note, rather than several notes demanding several subharmonics. This method included a speech/music discriminator that keyed on the monophonic nature of speech in most stereo programs. The point was to prevent male speech from generating subharmonics (though subharmonic synthesis can dramatize speech; see Project No. G279).

Compared to wideband octave division, subharmonic synthesis focuses on such a narrow band that pre- and especially post-filtering present relatively easy tasks. Dynamic tracking approaches 100% because subharmonic synthesizers rely on indirect division.

Beginner's View

Q. Nice summary, but I was hoping for more specific information on cooking up divider boxes and subharmonic synthesizers.

A. We'll be as specific as we can; but working up these effects demands access to an oscilloscope. You cannot assess critical phenomena without seeing the waveforms. This holds particularly for indirect dividers, where switching transitions must coincide with specific points on the sinewave.

Q. Assume I've got a scope. How do I work up an octave divider for guitar?

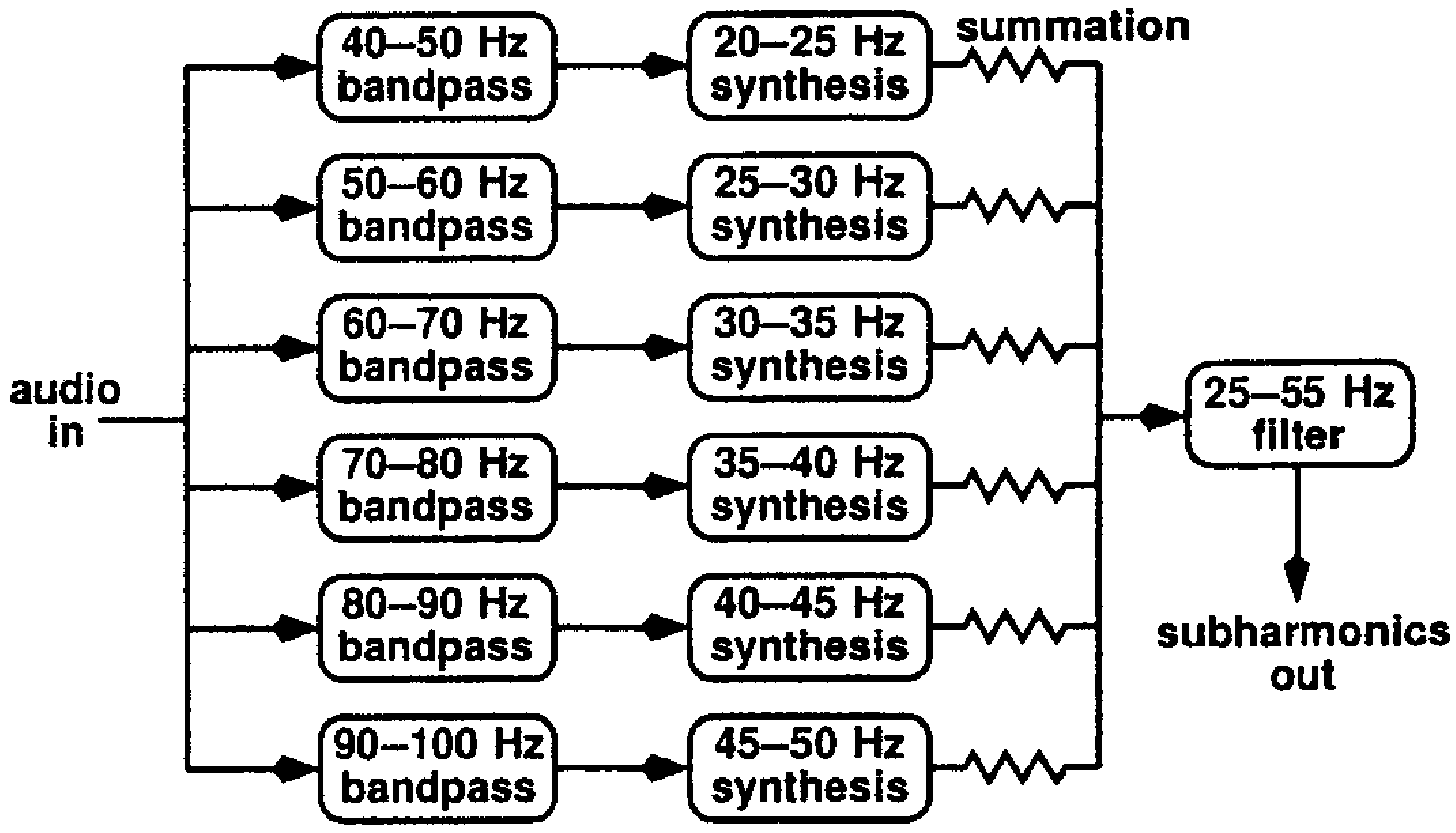
A. Refer to Fig. 6, a block diagram of the elements in a direct-divider box. The first step is to design an input filter that resists octave skipping.

Q. You're assuming that I know plenty about filters—which I don't.

A. Keep a copy of Ref. 6 handy. You can jumpstart the process by breadboarding the preamp, filter, and SSC out of Divide-O-Matic (IC1-c/-d/-a, IC2-a/-b). Hook one scope probe to the SSC output, to see octave skipping when it occurs; hook the other probe to the filter output, to see how well it's removing harmonics. Set the axe's volume and tone pots all the way open. Cleanly pick each note from the open low-E string, up to the 12th fret on the B-string. You don't have to make string-by-string, fret-by-fret notes, but get a general idea of where octave skipping tends to occur. Move the scope probe to the preamp output, pick the weak frets again, and you'll see that the fundamental has little amplitude on these frets. Also, note the effect on octave skipping of changing treble control pot R35 and the axe's tone pot.

To see what happens without filtering, connect C4 directly to the output of IC1-c. You'll find octave skipping rampant, but especially bad on open strings and on the first five frets. As you move up the neck, the output of the bottom four (wound) strings tends more to resemble a sinewave at the fundamental frequency, explaining why this region is most resistant to skip-

Fig. 3. One approach to subharmonic synthesis in wideband music. Audio feed splits into six 10-Hz bandpasses, each generating a subharmonic, followed by summation, and lowpass filtering to remove the 3/2 components, before being mixed with the music signal. Block diagram adapted from Ref. 1.



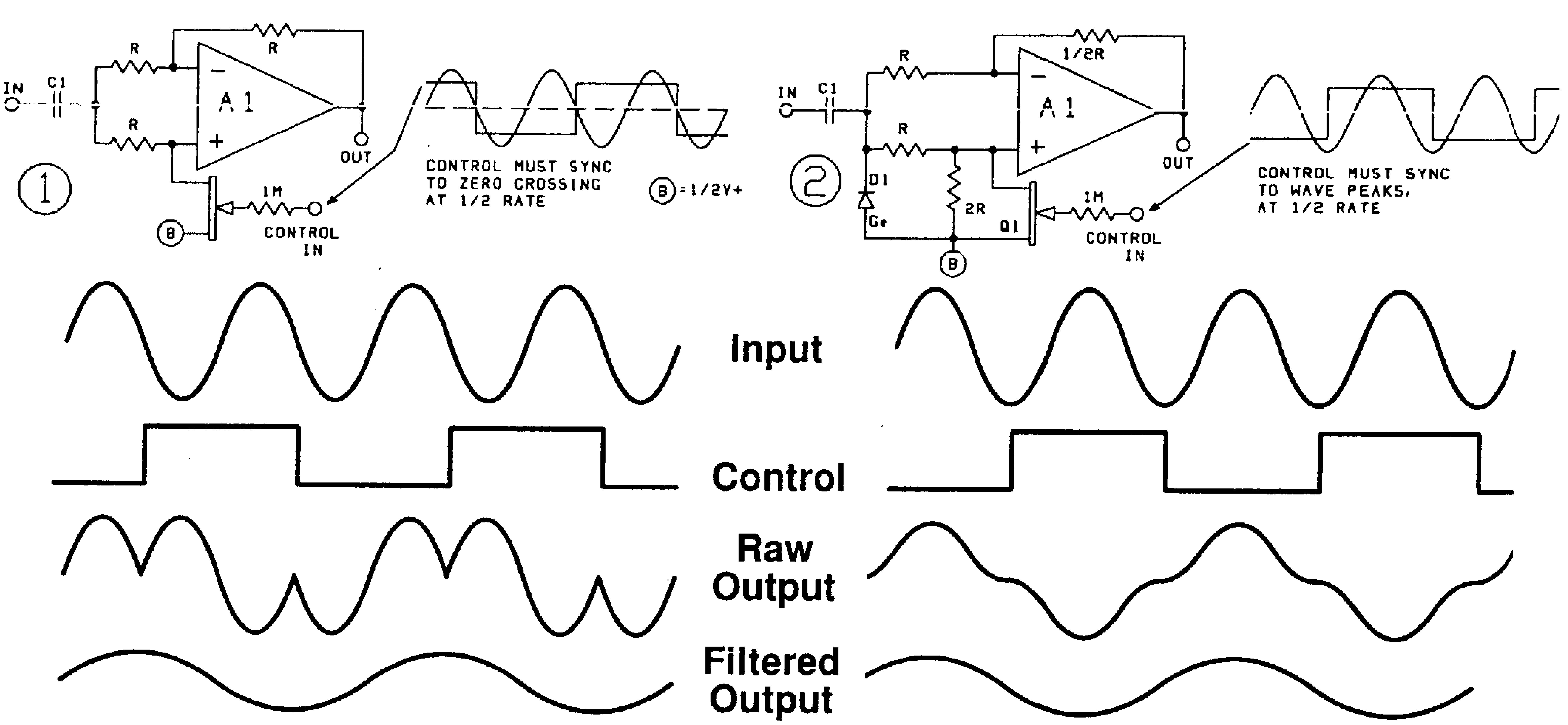


Fig. 4. Indirect divider circuits. 1—Op amp configured as a synchronous detector whose output equals sum and difference of inputs. If input is a sinewave of frequency x, and FET is toggled by a squarewave at a rate equal to 1/2x, then output contains [(x – 1/2x) = 1/2x]—i.e., a signal an octave below x—and [(x + 1/2x) = 3/2x]; also, many high harmonics of squarewave, but these are relatively easy to filter. Proper circuit function calls for squarewave transitions to coincide with sinewave zero-crossing points. 2—Function of this circuit is less obvious. Strategy is to slice, invert, scale, and apply a DC offset to the wave, which reassembles into a physical approximation of a tone one octave below the fundamental. Control transitions must coincide with sinewave peaks. Both approaches are more complicated than dividing the output of a sine-to-square converter, but track input dynamics and sound much smoother than squarewaves.

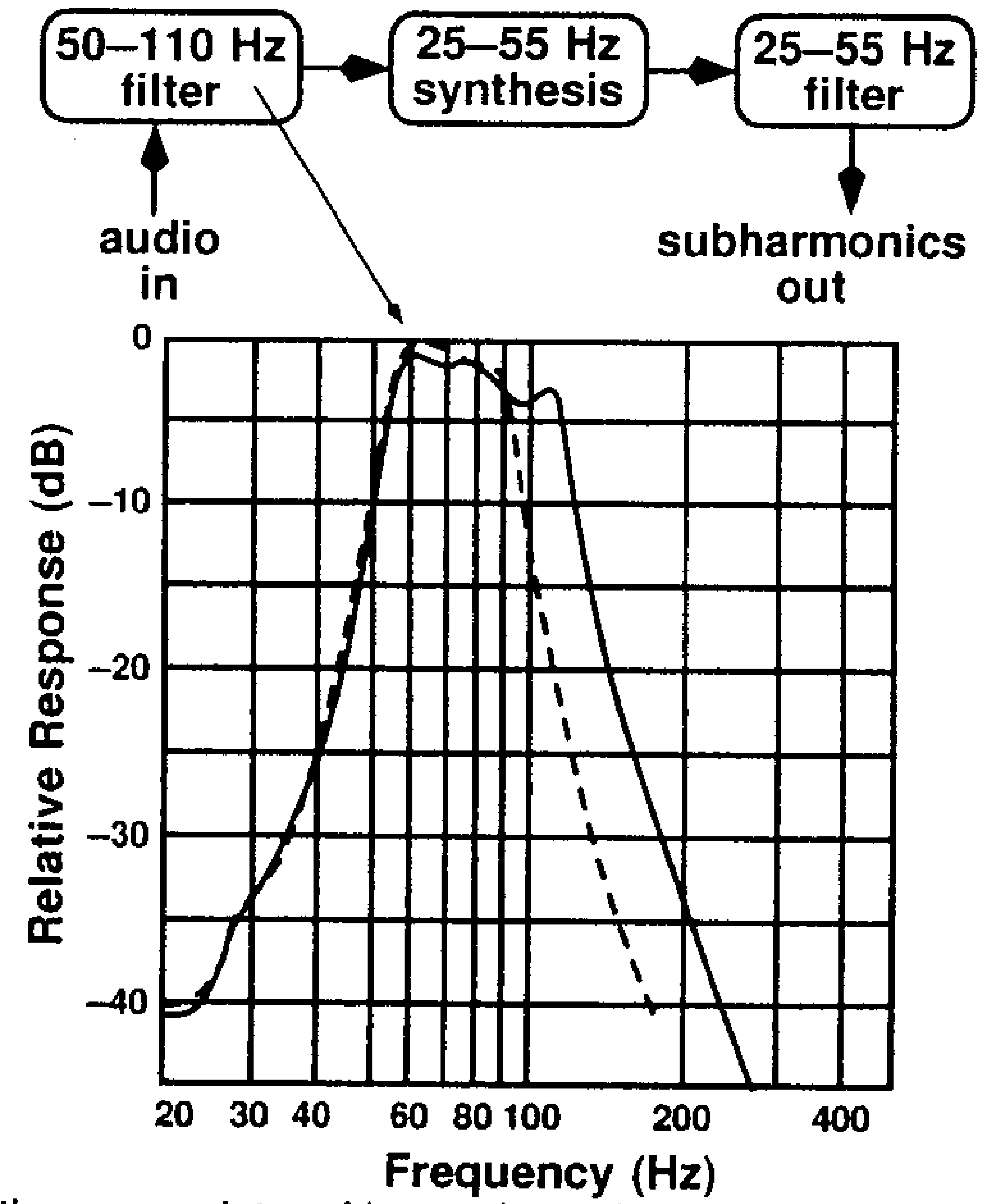


Fig. 5. Another approach to subharmonic synthesis in wideband music presumes that the deep bass usually contains only a single dominant note. Sharp bandpass isolates this note, from which a single subharmonic is derived. Graph shows two bandpass curves; solid line spans ~50–110 Hz; dotted-line graph has an upper limit of ~95 Hz, said to resist triggering on male voice in motion picture soundtracks. In both cases the single bandpass is realized by cascading three separate bandpasses. The downward slope within the passband is deliberate. Graphs redrawn after figures in Ref. 2.

ping.

This exercise demonstrates that octave skipping involves interactive variables, and shows why one filter won't render the whole fretboard skip-free. Preamp gain, filter type and slope, and comparator sensitivity all figure in the process. If you increase the filter slope by changing R35 and the axe's tone pot, the circuit becomes highly resistant to skipping, but the SSC fails to trigger as you move up the neck. Boosting the signal ahead of the comparator (R36) regains triggering, but may cause the SSC to oscillate while the axe is silent.

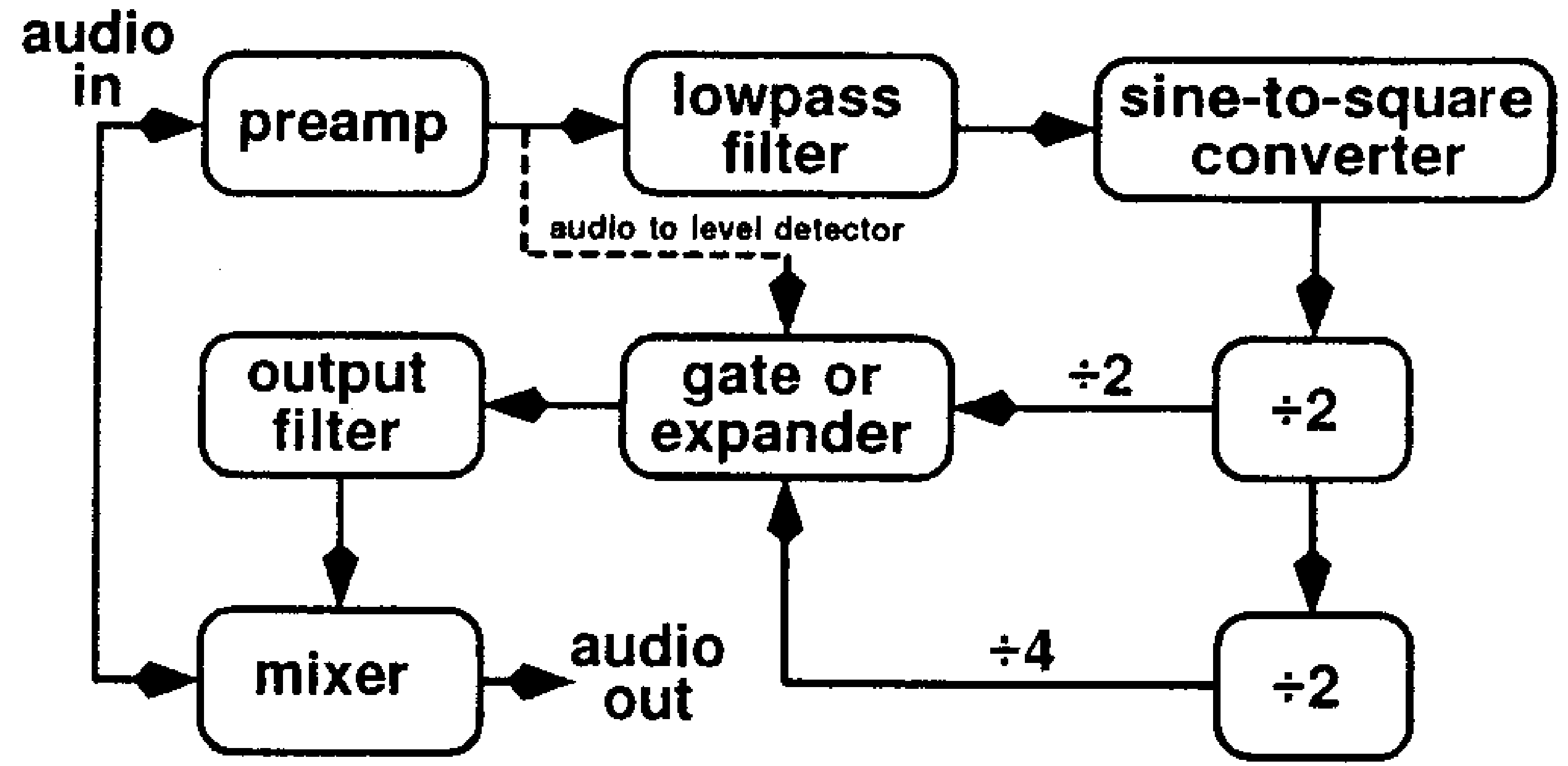


Fig. 6. Block diagram of basic direct-divider box for guitar. See text.

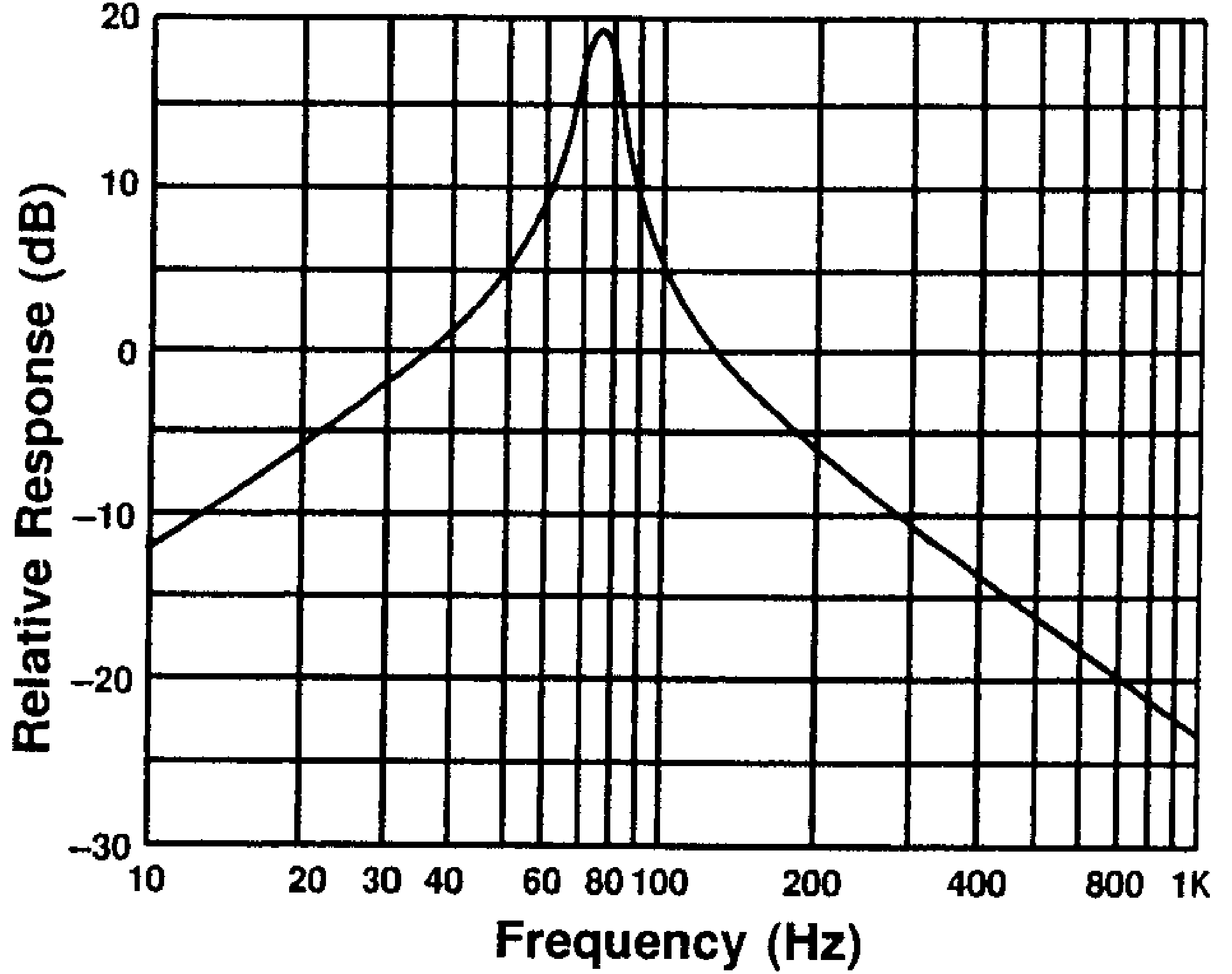


Fig. 7. Measured response of bandpass filter used in Divide-O-Matic (IC1-d, C18-19, R28-29-30).

Q. Why not include the high-E string in the tests?

A. Compromise. We're optimizing the filter to let us play over as much of the fretboard as possible, without octave skipping. Later, if you wish, you

(continued on page 12)

Project No. G277

Sub-O-Matic

Not a stompbox, but a fairly simple circuit that generates subharmonics for consumer stereo applications.

Circuit Function

The circuit has two identical signal paths; only the left is described. Line-level input couples through C18-R44 to voltage follower IC1-c, thence through R5 to noninverting summing amp IC1-a, whose output couples to a narrow bandpass filter comprised of IC2-b, -c, & -d, and their associated components. The bandpass covers ~45–95 Hz. Filter output couples through R32 to Schmitt trigger IC2-a, which acts as a zero-crossing detector. Q1 conditions IC2-a's output for reliable triggering of IC3, a CMOS divide-by-two block whose output toggles Q2 through R16.

IC2-d's output also couples to a synchronous detector comprised of IC4-b and the associated components. IC4-b's output contains frequencies equal to 1/2 and 3/2 the audio input. Lowpass filter IC4-c removes most of the 3/2 components. The filter's output couples to subharmonic level pot R45, which feeds a pair of noninverting amps (IC4-a and -d), which drive the left- and right-channel outputs.

Use

Turn R45 fully CCW. Connect unit's inputs to the line-level outputs of a consumer stereo device, such as a CD player, FM radio, or tape deck. Connect Sub-O-Matic's outputs to the line-level inputs of a stereo amplifier; establish desired listening level. Slowly turn R45 clockwise and note the increasing subharmonic level.

Notes

The prototype drew 3.5 ma running @7.8V. If the builder wishes to install a power switch, solder a SPST switch between the positive battery lead and the circuit board.

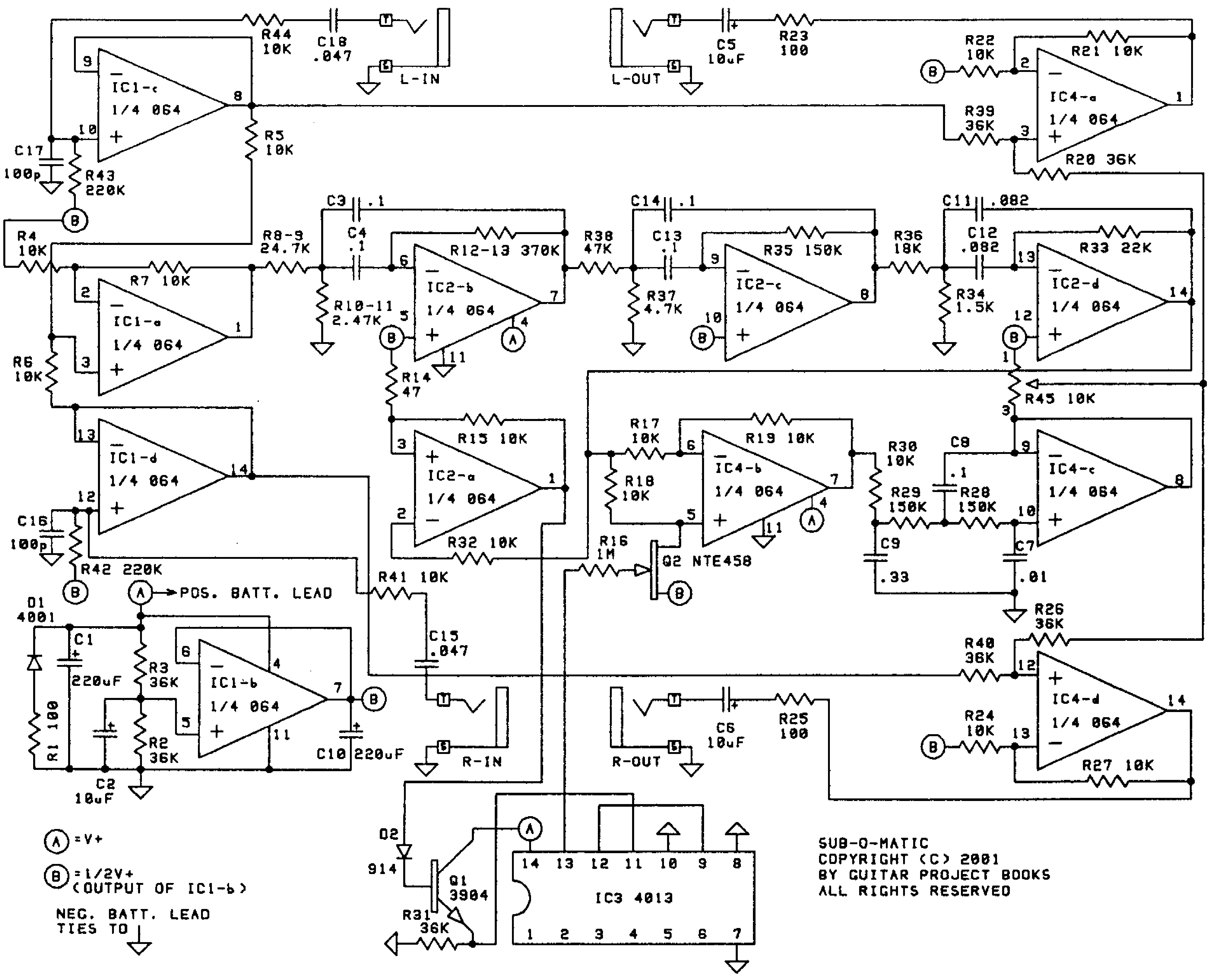
The three-stage bass bandpass filter is very sensitive to component values. C3,4,11,12,13,14 should measure within 2% of their stated values. This does not demand expensive 2% caps if a capacitance meter is available. Purchase 10 5% caps (such as Panasonic's B-series, available through Digi-Key); measure them, choose the ones closest to the nominal values. The parts used in the prototype measured within 1% of the stated values.

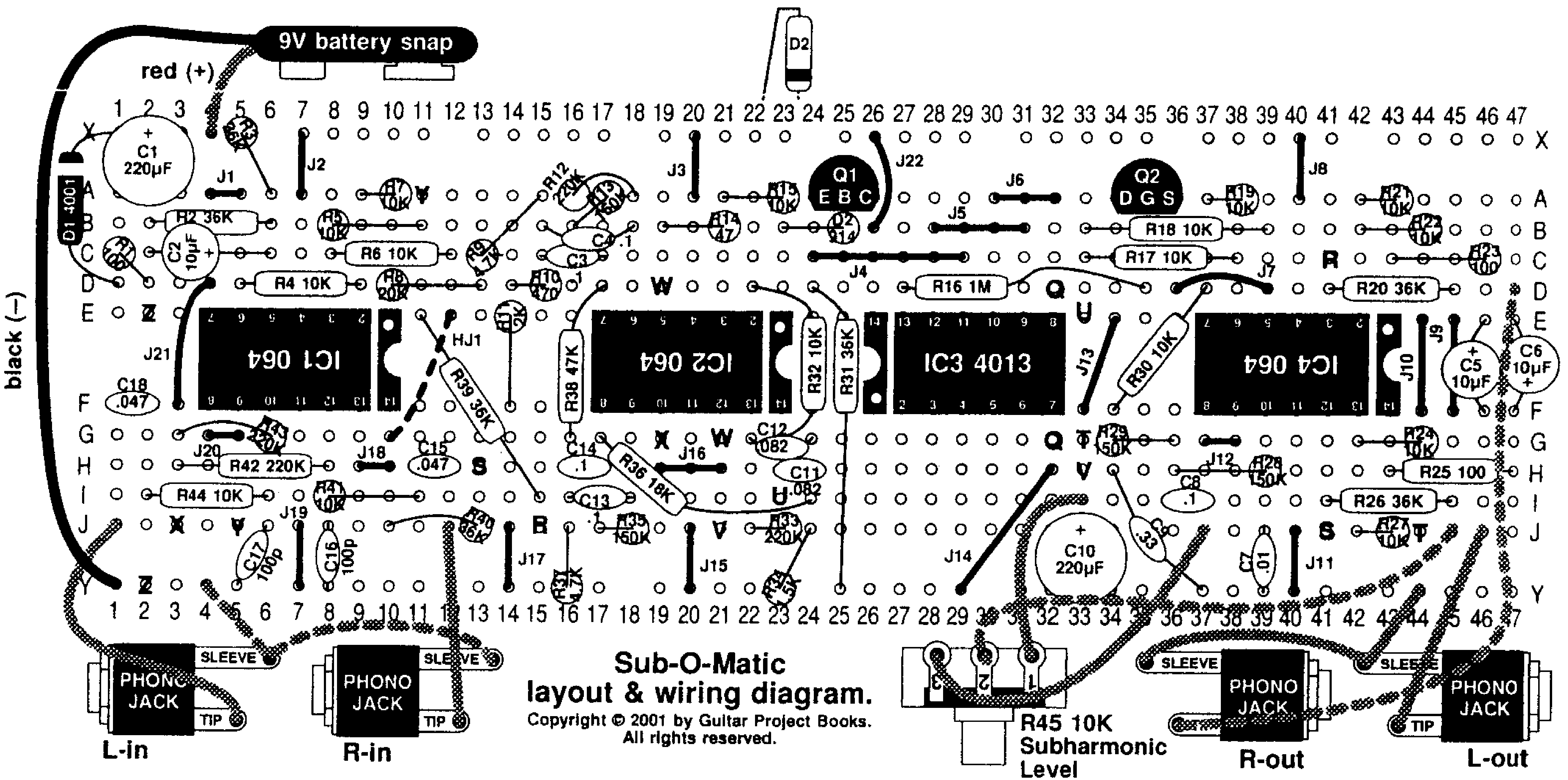
The resistances required for the first bandpass filter (IC2-b) are realized by summing common values (R8-R13). Like the caps, these resistances should measure within 2% of their stated values.

There is little point in using a subharmonic synthesizer with woofers much smaller than 8". In systems that lack a dedicated subwoofer, best results will be obtained from 10" or 12" woofers in sealed enclosures (sometimes called "infinite baffles"). Substantial enhancement will also be heard using full-size, wide-range headphones, but not with miniature phones that come with portable radios and CD players.

Sub-O-Matic's performance far exceeded the Editor's expectations. The unit enhanced both old/analog material, and modern/digital material that, presumably, had not been rolled off below 100 Hz.

Sub-O-Matic schematic.





Sub-O-Matic layout & wiring diagram.

Parts List/Soldering Checklist

Circuit Board Radio Shack p/n 276-170; Hosfelt p/n 42-183; or DC Electronics p/n J4-404

IC Sockets

- ☐ 14-pin for IC1; pin-1 to E10
- ☐ 14-pin for IC2; pin-1 to E23
- ☐ 14-pin for IC3; pin-1 to F26
- ☐ 14-pin for IC4; pin-1 to E43

Resistors

☐	R1	100	(brn-blk-brn)	C1-D2
☐	R2	36K	(org-blu-org)	B2-B6
☐	R3	36K	(org-blu-org)	X5-A6
☐	R4	10K	(brn-blk-org)	D5-D9
☐	R5	10K	(brn-blk-org)	B8-B11
☐	R6	10K	(brn-blk-org)	C8-C12
☐	R7	10K	(brn-blk-org)	A9-A10
☐	R8	20K	(red-blk-org)	D10-D13
☐	R9	4.7K	(yel-vio-red)	C13-A15
☐	R10	470	(yel-vio-brn)	D14-D15
☐	R11	2K	(red-blk-red)	E14-F14
☐	R12	220K	(red-red-yel)	A16-A18
☐	R13	150K	(brn-grn-yel)	B16-A17
☐	R14	47	(yel-vio-blk)	B19-B21
☐	R15	10K	(brn-blk-org)	A21-A23
☐	R16	1M	(brn-blk-grn)	D27-D35
☐	R17	10K	(brn-blk-org)	C33-C38
☐	R18	10K	(brn-blk-org)	B33-B39
☐	R19	10K	(brn-blk-org)	A37-A38
☐	R20	36K	(org-blu-org)	D41-D45
☐	R21	10K	(brn-blk-org)	A42-A43
☐	R22	10K	(brn-blk-org)	B42-B44
☐	R23	100	(brn-blk-brn)	C43-C46
☐	R24	10K	(brn-blk-org)	G42-G44
☐	R25	100	(brn-blk-brn)	H43-H47
☐	R26	36K	(org-blu-org)	I41-I45
☐	R27	10K	(brn-blk-org)	J42-J43
☐	R28	150K	(brn-grn-yel)	H36-H39
☐	R29	150K	(brn-grn-yel)	G34-G36
☐	R30	10K	(brn-blk-org)	D37-F34
☐	R31	36K	(org-blu-org)	D24-Y25
☐	R32	10K	(brn-blk-org)	D22-G23
☐	R33	220K	(red-red-yel)	J22-J23
☐	R34	1.5K	(brn-grn-red)	J24-Y23
☐	R35	150K	(brn-grn-yel)	J17-J18
☐	R36	18K	(brn-gry-org)	G17-I24
☐	R37	4.7K	(yel-vio-red)	J16-Y16
☐	R38	47K	(yel-vio-org)	D17-G16

☐ **[install hidden jumper HJ1 (E12-G10) before soldering R39]**

☐	R39	36K	(org-blu-org)	E11-I15
☐	R40	36K	(org-blu-org)	J10-J13
☐	R41	10K	(brn-blk-org)	I8-I11
☐	R42	220K	(red-red-yel)	H3-H8
☐	R43	220K	(red-red-yel)	G3-G6
☐	R44	10K	(brn-blk-org)	I2-I6

Bare Wire Jumpers

☐	J1	A4-A5
☐	J2	X7-A7
☐	J3	X20-A20
☐	J4	C24-C29
☐	J5	B28-B31
☐	J6	A30-A32
☐	J7	D36-D39
☐	J8	X40-A40
☐	J9	E45-F45
☐	J10	E44-F44
☐	J11	J40-Y40
☐	J12	G37-G38
☐	J13	E34-F33
☐	J14	H32-Y29
☐	J15	J20-Y20
☐	J16	H19-H21
☐	J17	J14-Y14
☐	J18	H9-H10
☐	J19	J7-Y7
☐	J20	G4-G5
☐	J21	D4-F3
☐	J22	X26-B26

Capacitors

☐	C1	220µF	X2-A2 (+ lead to X2)
☐	C2	10µF	C2-C6 (+ lead to C6)
☐	C3	0.1, 2%	C15-C17
☐	C4	0.1, 2%	B15-B18
☐	C5	10µF	E46-F46 (+ lead to E46)
☐	C6	10µF	E47-F47 (+ lead to F47)
☐	C7	0.01, 5%	J39-Y39
☐	C8	0.1, 5%	I36-I37
☐	C9	0.33, 5%	H34-Y37
☐	C10	220µF	J33-Y33 (+ lead to J33)
☐	C11	0.082, 2%	H23-H24
☐	C12	0.082, 2%	G22-G24
☐	C13	0.1, 2%	I16-I18
☐	C14	0.1, 2%	H16-H17
☐	C15	0.047	H11-H12
☐	C16	100pF	J8-Y8
☐	C17	100pF	J6-Y5
☐	C18	0.047	F1-F2

Flying Jumpers (<u>insulated</u> wire)

☐	QQ	D32-G32
☐	RR	J15-C41
☐	SS	H13-J41
☐	TT	G33-J44
☐	UU	I23-E33
☐	VV	J21-H33
☐	WW	D19-G21
☐	XX	J3-G19
☐	YY	J5-A11
☐	ZZ	E2-Y2

Semiconductors

☐	D1	1N4001	X1-D1 (banded end to X1)
☐	D2	1N914	B23-B25 (banded end to B25)
☐	IC1	TL064 quad op amp; pin-1 to E10	
☐	IC2	TL064 quad op amp; pin-1 to E23	
☐	IC3	4013; pin-1 to F26	
☐	IC4	TL064 quad op amp; pin-1 to E43	
☐	Q1	2N3904; emitter (E) to A24, base (B) to A25, collector (C) to A26	
☐	Q2	NTE458; drain (D) to A34, gate (G) to A35, source (S) to A36	

Potentiometers (T=terminal)

- ☐ R45 10K T1 to I33, T2 to J45; T3 to J37

Jacks (T=terminal)

- ☐ R-input jack (RCA/phono): tip to J12, sleeve to Y4
- ☐ L-input jack (RCA/phono): tip to J1, sleeve to Y4
- ☐ L-output jack (RCA/phono): tip to J46, sleeve to Y44
- ☐ R-output jack (RCA/phono): tip to D47, sleeve to Y44

9V Battery Leads

- ☐ black (–) lead to Y1
- ☐ red (+) lead to X4

Project No. G278

Sub-O-Matic II

This box takes bass guitar to the nether limit of human hearing.

Circuit Function

Axe feed couples through C10-R20 to voltage follower IC4-c, whose output couples through R19 to output buffer IC4-d; and also through C18-R33 to level detector amp IC1-c, whose output negotiates lowpass filter IC1-d, then traverses phase-shift network R30-C3. The boosted, filtered, and phase-shifted signal then couples through R28-C13 to variable gain amp IC2-a, thence through R27 to Schmitt trigger IC2-b, which acts as a zero-crossing detector. IC2-b's output drives buffer Q1 through D2 and R7; Q1's output feeds IC3, a CMOS flip-flop configured as a divide-by-2 circuit whose output drives Q2 through R8. Q2, D3, IC4-b, and the associated components form a frequency divider circuit similar to the one described in Figs. 4–2 & 7; R35 allows trimming to optimize divider action. The raw divider output feeds lowpass filter IC4-a. The filtered subharmonic signal couples to R37, which varies the level present in both mixed and raw subharmonic output.

Use

Switch and pots have these functions:

S1	effect/bypass
R35	frequency divider trim
R36	subharmonic sensitivity
R37	subharmonic output level

First, trim D3's bias point. Connect a 9V battery, insert axe plug into input jack to power-up circuit. Clip one lead of a voltmeter to jumper J19; the other lead to jumper J3. This measures the battery voltage; record the value obtained (which should fall in the range 7V to 10V). Divide this value by 2.128, then trim R35 to give the calculated value at R35's wiper (accessible at D3's cathode lead, or at the '+' lead of C11). For example, a supply voltage of 8.22 volts means that the voltage at R35's wiper should be trimmed to (8.22 ÷ 2.128) = 3.86V.

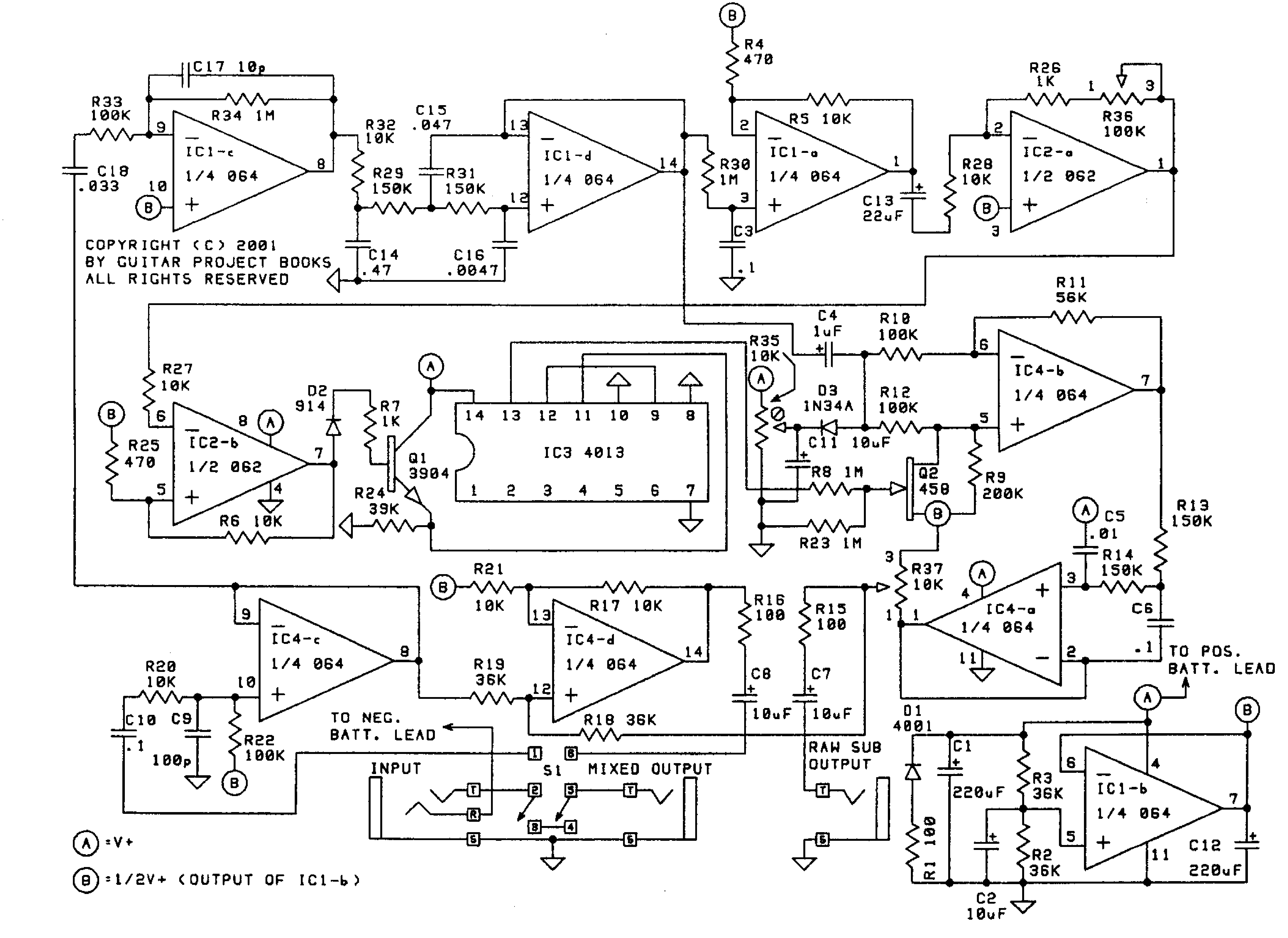
Turn R37 fully CCW, set R36 at about 9 o'clock. Connect unit to axe and amp, initially using the mixed output; establish desired listening level. In this condition the box acts as a unity-gain, noninverting buffer. Slowly turn R37 clockwise and note the increasing subharmonic level in the output. Adjust R36 if necessary to alter the subharmonic generation threshold.

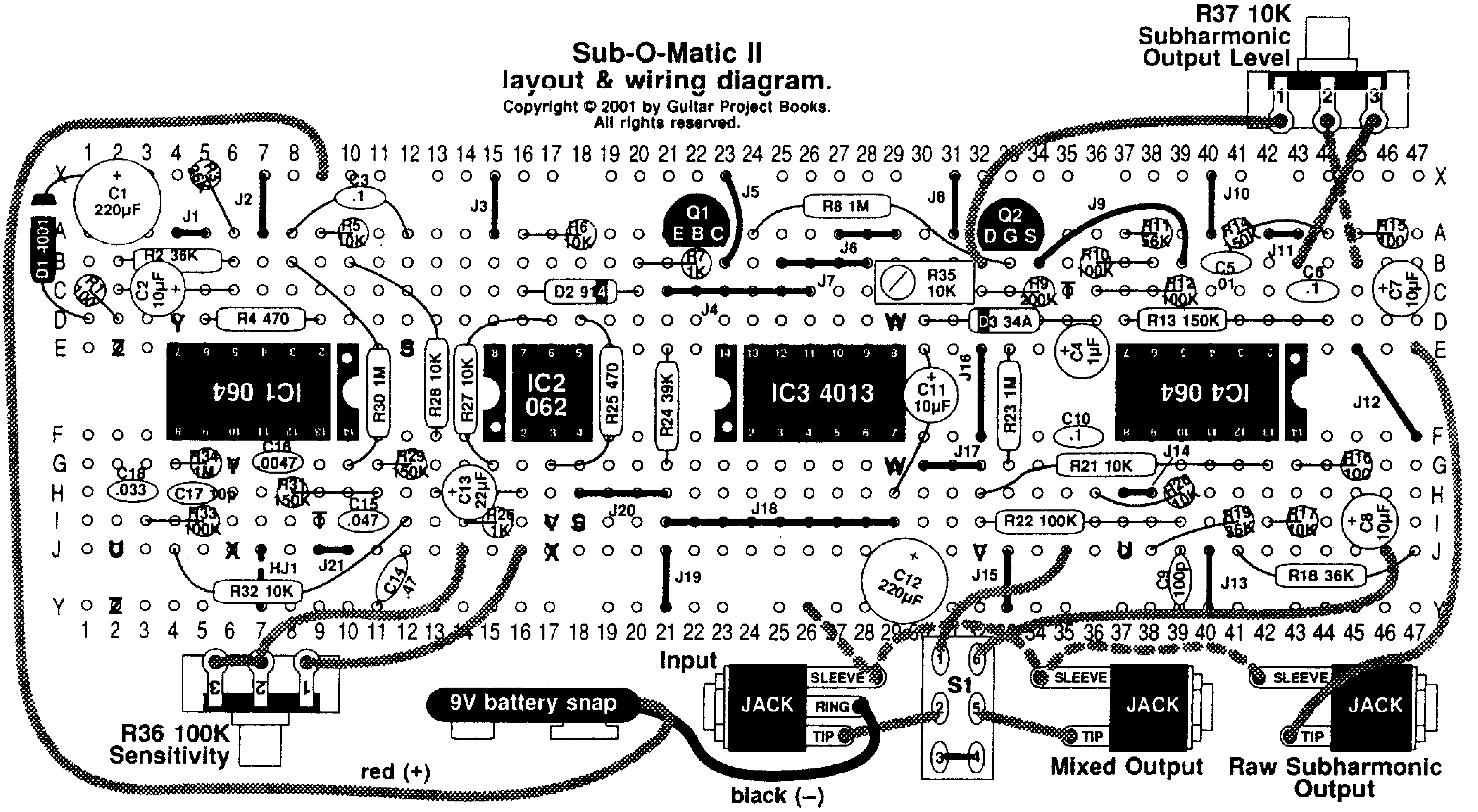
Notes

Sub-O-Matic II is designed to respond to signals in the range 40–80 Hz, corresponding to the open low-E, up to the second fret on the D-string of a conventionally tuned four-string bass guitar. There is little point in building this box unless the player has access to an amp/speaker system with substantial response below 40 Hz; or unless the player plans to record directly from the box, on suitably wideband media. The raw subharmonic output is provided to facilitate use of a dedicated subwoofer system.

The prototype drew about 3 milliamps. A metal box is recommended to shield several high-impedance nodes from 60 Hz power fields, which could cause the circuit to generate a spurious 30 Hz tone.

Sub-O-Matic II schematic.





Sub-O-Matic II layout & wiring diagram.

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Parts List/Soldering Checklist

Circuit Board Radio Shack p/n 276-170; Hosfelt p/n 42-183; or DC Electronics p/n J4-404

IC Sockets

[ ] 14-pin for IC1; pin-1 to E10
[ ] 8-pin for IC2; pin-1 to F15
[ ] 14-pin for IC3; pin-1 to F23
[ ] 14-pin for IC4; pin-1 to E43

Resistors

[ ] R1 100 (brn-blk-brn) C1-D2
[ ] R2 36K (org-blu-org) B2-B6
[ ] R3 36K (org-blu-org) X5-A6
[ ] R4 470 (yel-vio-brn) D5-D9
[ ] R5 10K (brn-blk-org) A9-A10
[ ] R6 10K (brn-blk-org) A16-A18
[ ] R7 1K (brn-blk-red) B20-B22
[ ] R8 1M (brn-blk-grn) A24-B33
[ ] R9 200K (red-blk-yel) C32-C34
[ ] R10 100K (brn-blk-yel) B36-B38
[ ] R11 56K (grn-blu-org) A37-A38
[ ] R12 100K (brn-blk-yel) C36-C39
[ ] R13 150K (brn-grn-yel) D37-D44
[ ] R14 150K (brn-grn-yel) A41-A44
[ ] R15 100 (brn-blk-brn) A45-A46
[ ] R16 100 (brn-blk-brn) G43-G45
[ ] R17 10K (brn-blk-org) I42-I43
[ ] R18 36K (org-blu-org) J41-J47
[ ] R19 36K (org-blu-org) J38-I41
[ ] R20 10K (brn-blk-org) H36-H39
[ ] R21 10K (brn-blk-org) H32-G42
[ ] R22 100K (brn-blk-yel) I32-I39
[ ] R23 1M (brn-blk-grn) E33-G33
[ ] R24 39K (org-wht-org) E21-G21
[ ] R25 470 (yel-vio-brn) D18-G17
[ ] R26 1K (brn-blk-red) I14-I15
[ ] R27 10K (brn-blk-org) D17-G15
[ ] R28 10K (brn-blk-org) B10-F13
[ ] R29 150K (brn-grn-yel) G11-G12
[ ] R30 1M (brn-blk-grn) B8-G10
[ ] R31 150K (brn-grn-yel) H8-H11
[ ] [install hidden jumper HJ1 (J7-Y7) before soldering R32)
[ ] R32 10K (brn-blk-org) J4-I12
[ ] R33 100K (brn-blk-yel) I3-I5
[ ] R34 1M (brn-blk-grn) G4-G5

Bare Wire Jumpers

[ ] J1 A4-A5
[ ] J2 X7-A7
[ ] J3 X15-A15
[ ] J4 C21-C26
[ ] J5 X23-B23
[ ] J6 A27-A29
[ ] J7 B25-B28
[ ] J8 X31-A31
[ ] J9 B34-B39
[ ] J10 X40-A40
[ ] J11 A42-A43
[ ] J12 E45-F47
[ ] J13 J40-Y40
[ ] J14 H37-H38
[ ] J15 J33-Y33
[ ] J16 E32-F32
[ ] J17 G30-G32
[ ] J18 I21-I29
[ ] J19 J21-Y21
[ ] J20 H18-H21
[ ] J21 J9-J10

Capacitors

[ ] C1 220µF X2-A2 (+ lead to X2)
[ ] C2 10µF C2-C6 (+ lead to C6)
[ ] C3 0.1 A8-A12
[ ] C4 1µF E35-E36 (+ lead to E35)
[ ] C5 0.01, 5% B40-B41
[ ] C6 0.1, 5% C43-C44
[ ] C7 10µF C46-C47 (+ lead to C46)
[ ] C8 10µF I45-I46 (+ lead to I45)
[ ] C9 100pF J39-Y39
[ ] C10 0.1 F35-F36
[ ] C11 10µF F30-H29 (+ lead to F30)
[ ] C12 220µF J30-Y29 (+ lead to J30)
[ ] C13 22µF H13-H16 (+ lead to H13)
[ ] C14 0.47, 5% J12-Y11
[ ] C15 0.047, 5% I10-I11
[ ] C16 0.0047, 5% G7-G8
[ ] C17 10pF H4-H5
[ ] C18 0.033 H2-H3

Flying Jumpers (insulated wire)

[ ] SS E12-I18
[ ] TT I9-C35
[ ] UU J2-J37
[ ] VV I17-J32
[ ] WW D29-G29
[ ] XX J6-J17
[ ] YY D4-G6
[ ] ZZ E2-Y2

Semiconductors

[ ] D1 1N4001 X1-D1 (banded end to X1)
[ ] D2 1N914 C16-C20 (banded end to C20)
[ ] D3 1N34A D30-D36 (banded end to D30)
[ ] IC1 TL064 quad op amp; pin-1 to E10
[ ] IC2 TL062 dual op amp; pin-1 to F15
[ ] IC3 4013; pin-1 to F23
[ ] IC4 TL064 quad op amp; pin-1 to E43
[ ] Q1 2N3904; emitter (E) to A21, base (B) to A22, collector (C) to A23
[ ] Q2 NTE458; drain (D) to A32, gate (G) to A33, source (S) to A34

Trimpot

[ ] R35 10K multiturn trimpot; terminals go in C29-30-31

Potentiometers (T=terminal)

[ ] R36 100K T1 to J16; T2 to T3 and to J14
[ ] R37 10K T1 to B32, T2 to B45, T3 to B43

Jacks (T=terminal)

[ ] input jack (1/4" 3-terminal/stereo): tip to T2 of S1; ring to black (–) battery lead, sleeve to Y26
[ ] mixed output jack (1/4" 2-terminal/mono): tip to T5 of S1, sleeve to Y26
[ ] raw subharmonic output jack (1/4" 2-terminal/mono): tip to E47, sleeve to Y26

Switches (T=terminal)

[ ] S1 (DPDT stomp switch): T1 to J35; T2 to tip of input jack; T3 to T4; T5 to tip of mix output jack; T6 to J46

9V Battery Leads

[ ] black (–) lead to ring of input jack
[ ] red (+) lead to X9

Project No. G279
Sub-O-Matic III

Subharmonic enhancement for voice, with selectable frequency ranges.

Circuit Function

The line-level feed from a microphone preamp couples through C12-R23 to voltage follower IC4-c, whose output couples through R22 to noninverting output amp IC4-d; and through C20-R39 to inverting amp IC1-c, whose gain is fixed at 10. IC1-c's output feeds two separate bandpass filters, one consisting of IC1-a and -d; the other consisting of IC2-b and -c; the former filter spans ~100–240 Hz, the latter ~50–120 Hz. Each filter's output couples to one throw of S2-a, whose center pole feeds a zero-crossing detector comprised of IC2-a and -d (R41 alters the sensitivity of the zero-crossing detector); and also feeds a synchronous detector, comprised of IC4-b, Q1, and the associated components. The output of the zero-crossing detector couples through D2 to IC3, a CMOS divide-by-two circuit whose output drives Q1 through R7. The raw output of IC4-b couples to a lowpass filter comprised of IC4-a and associated components, and segments b and c of S2. The net effect of switching S2 is to select which bandpass filter feeds the synchronous detector, and to alter the frequency range of the lowpass filter that receives the detector's output. Pot R42 varies both the raw subharmonic output level and the subharmonic level of the mixed output.

Use

Pots and switches have these functions:

R41	sensitivity
R42	subharmonic output level
S1	effect/bypass
S2	band select low/high

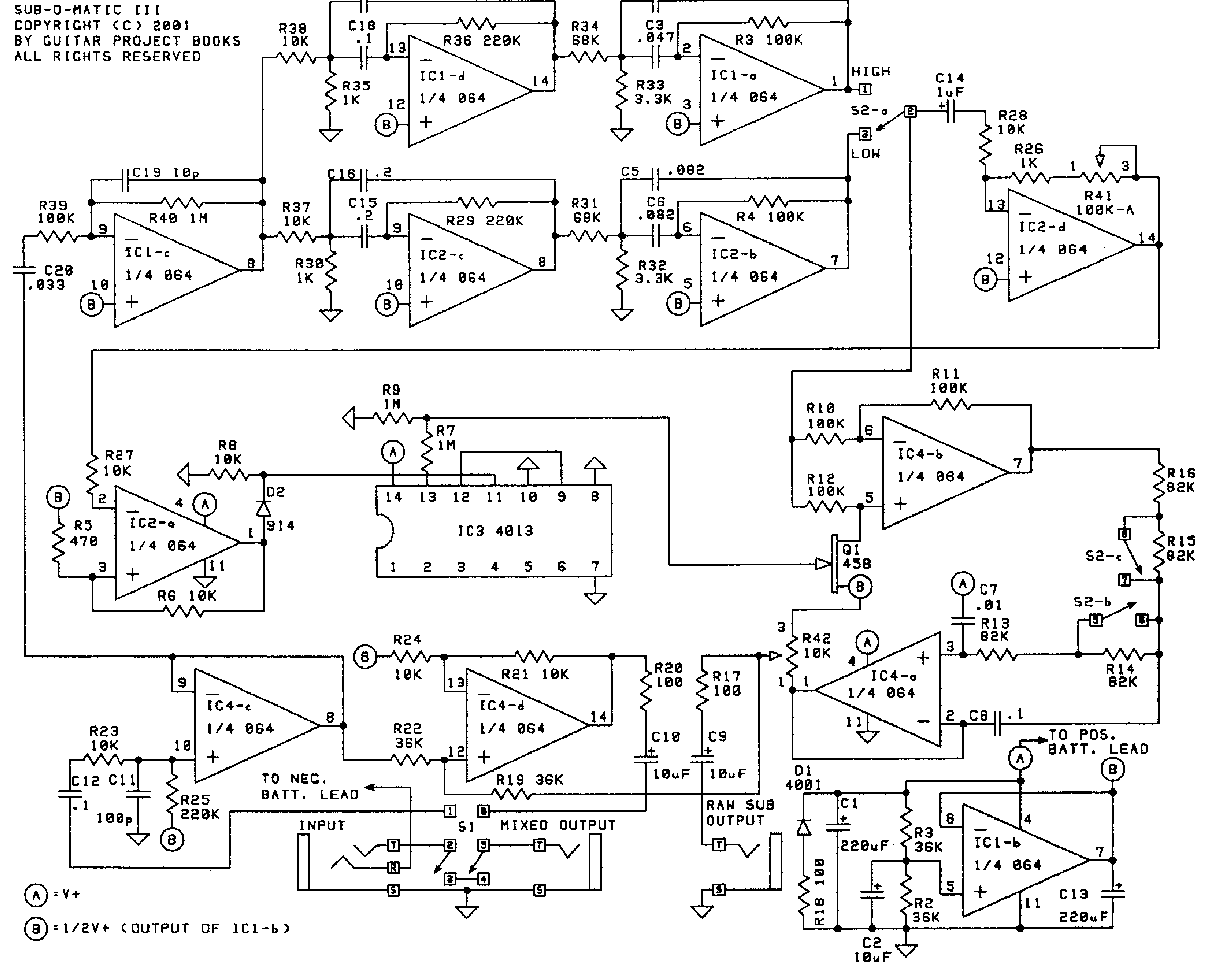
Initial settings: R41 10 o'clock, R42 fully CCW, S1 effect in, S2 in low position. Connect unit's input to line-level output of a microphone preamp, connect unit's output to amplifier driving wide-range headphones, or speakers whose response extends below 40 Hz. Establish desired listening level. While a deep male voice speaks into the microphone, slowly turn R42 clockwise and note the rising subharmonic component. Adjust R41 if necessary to achieve subharmonic synthesis. Depending on pitch of the voice, it may be necessary to toggle S2 to the high range.

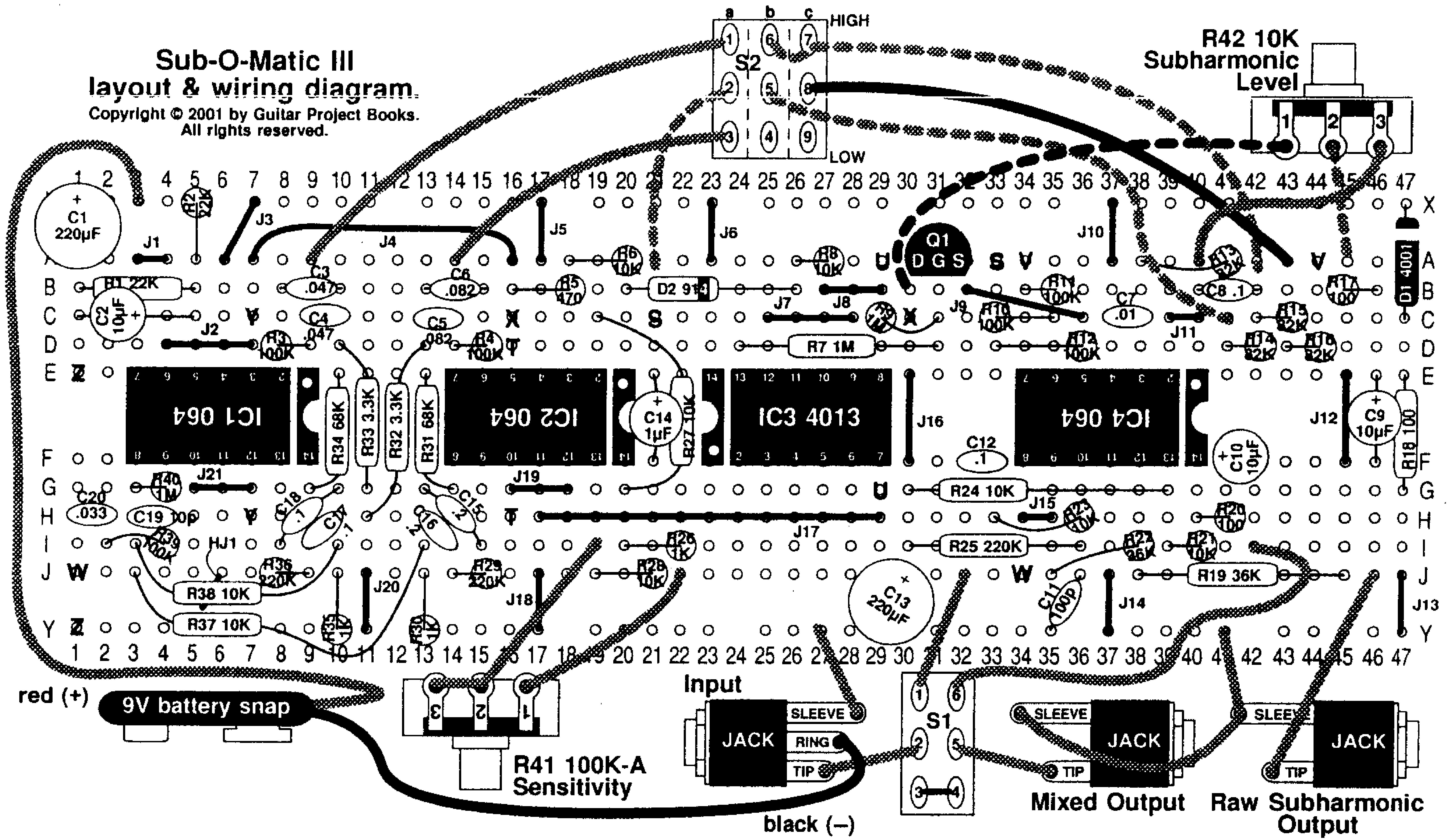
Notes

The prototype drew ~4.2 ma @ 7.8V.

The low and high bandpass filters were optimized, respectively, for deep male and female voices. The builder wishing to shift the frequency range(s) can do so by scaling the filter caps (C3,4,5,6,7,8,15,16,17,18) up or down by the same percentage.

Sub-O-Matic III schematic.





Sub-O-Matic III layout & wiring diagram.

Parts List/Soldering Checklist

Circuit Board Radio Shack p/n 276-170; Hosfelt p/n 42-183; or DC Electronics p/n J4-404

IC Sockets

[ ] 14-pin for IC1; pin-1 goes in E9
[ ] 14-pin for IC2; pin-1 goes in E20
[ ] 14-pin for IC3; pin-1 goes in F23
[ ] 14-pin for IC4; pin-1 goes in E40

Resistors

[ ] R1 22K (red-red-org) B1-B5
[ ] R2 22K (red-red-org) X5-A5
[ ] R3 100K (brn-blk-yel) D8-D9
[ ] R4 100K (brn-blk-yel) D14-D15
[ ] R5 470 (yel-vio-brn) B16-B18
[ ] R6 10K (brn-blk-org) A18-A20
[ ] R7 1M (brn-blk-grn) D24-D31
[ ] R8 10K (brn-blk-org) A26-A27
[ ] R9 1M (brn-blk-grn) C29-C31
[ ] R10 100K (brn-blk-yel) C33-C35
[ ] R11 100K (brn-blk-yel) B34-B35
[ ] R12 100K (brn-blk-yel) D33-D36
[ ] R13 82K (gry-red-org) A38-A41
[ ] R14 82K (gry-red-org) D41-D42
[ ] R15 82K (gry-red-org) C42-C43
[ ] R16 82K (gry-red-org) D43-D44
[ ] R17 100 (brn-blk-brn) B45-B46
[ ] R18 100 (brn-blk-brn) E47-G47
[ ] R19 36K (org-blu-org) J38-J45
[ ] R20 100 (brn-blk-brn) H40-H41
[ ] R21 10K (brn-blk-org) I39-I40
[ ] R22 36K (org-blu-org) J35-I38
[ ] R23 10K (brn-blk-org) H33-H36
[ ] R24 10K (brn-blk-org) G30-G39
[ ] R25 220K (red-red-yel) I30-I36
[ ] R26 1K (brn-blk-red) I20-I22
[ ] R27 10K (brn-blk-org) C19-G20
[ ] R28 10K (brn-blk-org) J19-J21
[ ] R29 220K (red-red-yel) J14-J15
[ ] R30 1K (brn-blk-red) J13-Y13
[ ] R31 68K (blu-gry-org) E13-G14
[ ] R32 3.3K (org-org-red) D13-H11
[ ] R33 3.3K (org-org-red) D10-G11
[ ] R34 68K (blu-gry-org) E10-G9
[ ] R35 1K (brn-blk-red) J10-Y10
[ ] R36 220K (red-red-yel) J8-J9

[install hidden jumper HJ1 (J6-Y5) before soldering R37 & R38]

[ ] R37 10K (brn-blk-org) J3-I13
[ ] R38 10K (brn-blk-org) I3-I10
[ ] R39 100K (brn-blk-yel) I2-I4
[ ] R40 1M (brn-blk-grn) G3-G4

Bare Wire Jumpers

[ ] J1 A3-A4
[ ] J2 D4-D7
[ ] J3 X7-A6
[ ] J4 A7-A16
[ ] J5 X17-A17
[ ] J6 X23-A23
[ ] J7 C25-C28
[ ] J8 B27-B29
[ ] J9 B32-C36
[ ] J10 X37-A37
[ ] J11 C39-C40
[ ] J12 E45-F45
[ ] J13 J47-Y47
[ ] J14 J37-Y37
[ ] J15 H34-H35
[ ] J16 E30-F30
[ ] J17 H17-H29
[ ] J18 J17-Y17
[ ] J19 G16-G18
[ ] J20 J11-Y11
[ ] J21 G5-G7

Capacitors

[ ] C1 220µF X1-A1 (+lead to X1)
[ ] C2 10µF C1-C5 (+lead to C5)
[ ] C3 0.047, 5% B8-B10
[ ] C4 0.047, 5% C9-C10
[ ] C5 0.082, 5% C13-C14
[ ] C6 0.082, 5% B13-B15
[ ] C7 0.01, 5% C37-C38
[ ] C8 0.1, 5% B40-B42
[ ] C9 10µF E46-F46 (+lead to E46)
[ ] C10 10µF F41-F42 (+lead to F41)
[ ] C11 100pF J36-Y35
[ ] C12 0.1 F32-F33
[ ] C13 220µF J30-Y29 (+lead to J30)
[ ] C14 1µF E21-F21 (+lead to E21)
[ ] C15 0.2.5% G13-I15
[ ] C16 0.2, 5% H13-I14
[ ] C17 0.1, 5% H10-I9
[ ] C18 0.1, 5% G10-I8
[ ] C19 10pF H3-H4
[ ] C20 0.033 H1-H2

Flying Jumpers (insulated wire)

[ ] SS C21-A33
[ ] TT D16-H16
[ ] UU A29-G29
[ ] VV A34-A44
[ ] WW J1-J34
[ ] XX C16-C30
[ ] YY C7-H7
[ ] ZZ E1-Y1

Semiconductors

[ ] D1 1N4001 X47-C47 (banded end to X47)
[ ] D2 1N914 B20-B26 (banded end to B26)
[ ] IC1 TL064 quad op amp; pin-1 to E9
[ ] IC2 TL064 quad op amp; pin-1 to E20
[ ] IC3 4013; pin-1 to F23
[ ] IC4 TL064 quad op amp; pin-1 to E40
[ ] Q1 NTE458 drain (D) to A30, gate (G) to A31, source (S) to A32

Potentiometers (T=terminal)

[ ] R41 100K audio-taper T1 to J22, T2 to T3 and to I19
[ ] R42 10K T1 to B30, T2 to A45, T3 to A40

Jacks (T=terminal)

[ ] input jack (1/4" 3-terminal/stereo): tip to T2 of S1, ring to negative (–) battery lead, sleeve to Y27
[ ] mixed output jack (1/4" 2-terminal/mono): tip to T5 of S1, sleeve to Y41
[ ] raw subharmonic output jack (1/4" 2-terminal mono): tip to J46, sleeve to Y41

Switches (T=terminal, NC=no connection)

[ ] S1 (DPDT stomp switch): T1 to J32; T2 to tip of input jack; T3 to T4; T5 to tip of mixed output jack; T6 to I42
[ ] S2 (3PDT slide or toggle switch): T1 to A9, T2 to A21, T3 to A14, T4 NC, T5 to C41, T6 to T7 and to A42, T8 to A43, T9 NC

9V Battery Leads

[ ] black (–) lead to ring of input jack
[ ] red (+) lead to X3

Project No. G284

Divide-O-Matic

Octave divider as distortion box.

Circuit Function

Signal Path: Axe feed couples through C20-R33 to inverting preamp IC1-c, whose gain is fixed at 10. Preamp output negotiates 78-Hz bandpass filter IC1-d, then passes to variable lowpass filter IC1-a, thence to IC2-b, an inverting amp whose gain varies under control of R36. IC2-b output feeds comparator IC2-a, whose squarewave output couples through D2 to IC3, a 4013 configured as two sequential divide-by-two circuits. The ÷2 signal couples through R9 to pot R37, thence through R11-C5 to the input of a tone network comprised of IC4-b and associated components. The ÷4 signal couples through R23 to pot R38, thence through R22-C12 to the input of the tone network. Dual pot R39 is wired to provide bass boost/treble cut when CCW; bass cut/treble boost when CW.

The output of IC1-c also couples to pot R40, whose wiper ties through R16 to the inverting input of IC4-b. R40 varies the percentage of clean signal in the output mix.

IC4-b's output couples through C9-R18 to the output path.

Gate Control Path: IC1-c's output also couples through C11-R20 to IC1-a, an inverting amp whose gain varies under control of R41. IC4-a's output couples through D3 to integrator/auto-variable decay circuit IC2-d, whose output ties through R24 to comparator IC2-c. When an audio feed is present, the output of IC2-c flips high, turning Q1 off through R26. This allows IC2-b to function normally. When the signal level drops below threshold, the output of IC2-c flips low, turning Q1 on, reducing IC2-b's gain to 0, thereby muting the divider path.

Use

Pots and switch have these functions:

R35	treble sensitivity
R36	SSC sensitivity
R37	÷2 output level
R38	÷4 output level
R39	tone
R40	clean output level
R41	gate sensitivity
S1	effect/bypass

Initial control settings: S1 effect in; R37, R38 fully CCW; R35, R39 centered; R36, R40, R41 about 10 o'clock. Connect unit to axe and amp, establish desired listening level. In this state the box acts as a noninverting preamp. Slowly advance R37 until a ÷2 signal is clearly heard. Take the tone pot R39 through its range and note the effect on sound. Return R37 fully CCW, then turn R38 CW until a ÷4 signal is heard; again take the tone pot through its range.

Notes

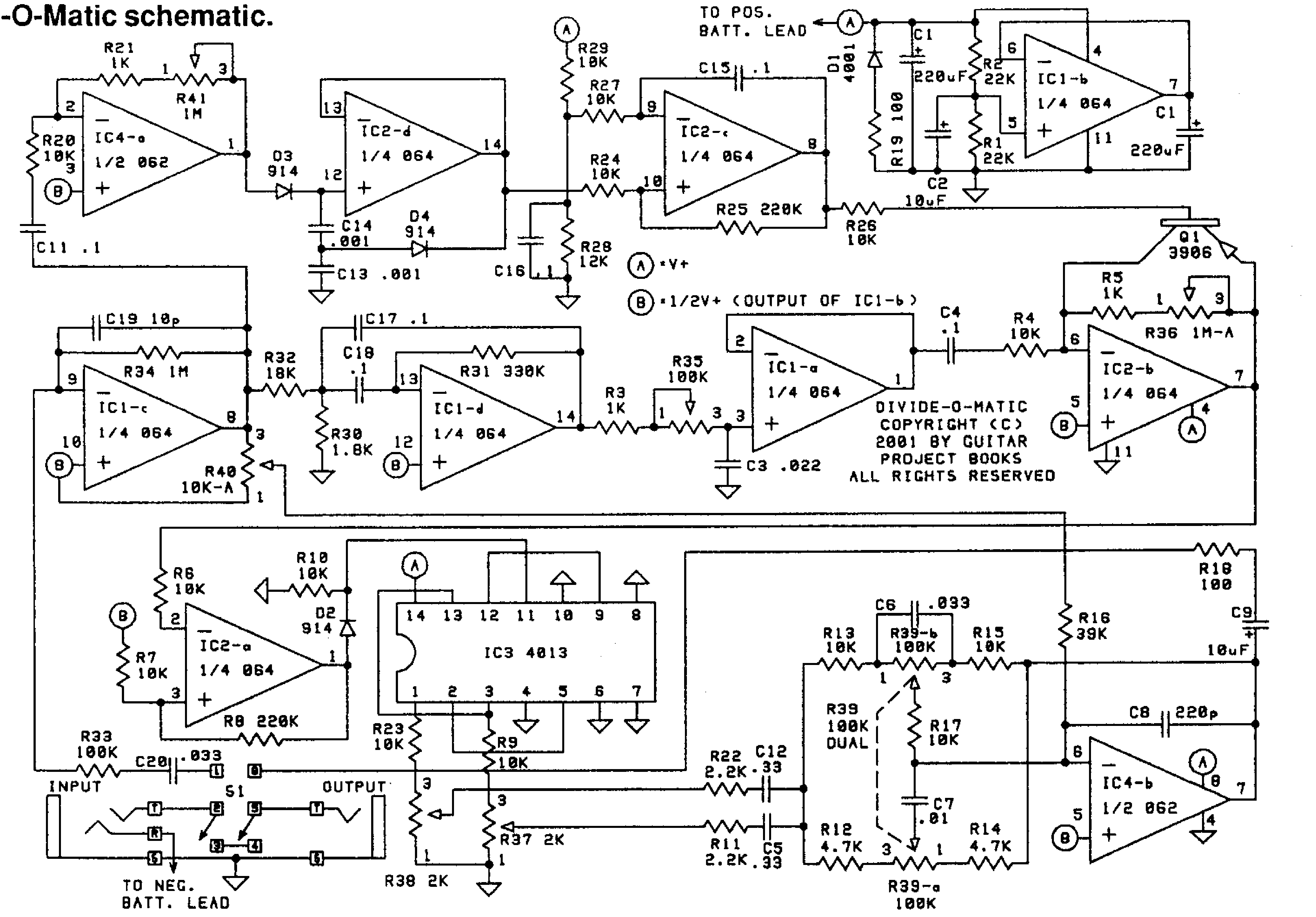
Divide-O-Matic exhibits two distinct response patterns at two extremes of control settings. The first settings consist of: treble sensitivity minimum, SSC threshold about 10 o'clock, axe tone pot fully off. In this state the box is extremely resistant to octave skipping from the open low-E string up to about the 10th fret on the D-string, dead frets excepted.

The second settings consist of: treble sensitivity and SSC sensitivity maximum, axe tone pot fully open. In this state the box is highly resistant to octave skipping on all strings *at and above the 12th fret.* While little sustain is possible on the high-E string in this region, brief notes are playable. Sustain increases to several seconds on the heavier strings.

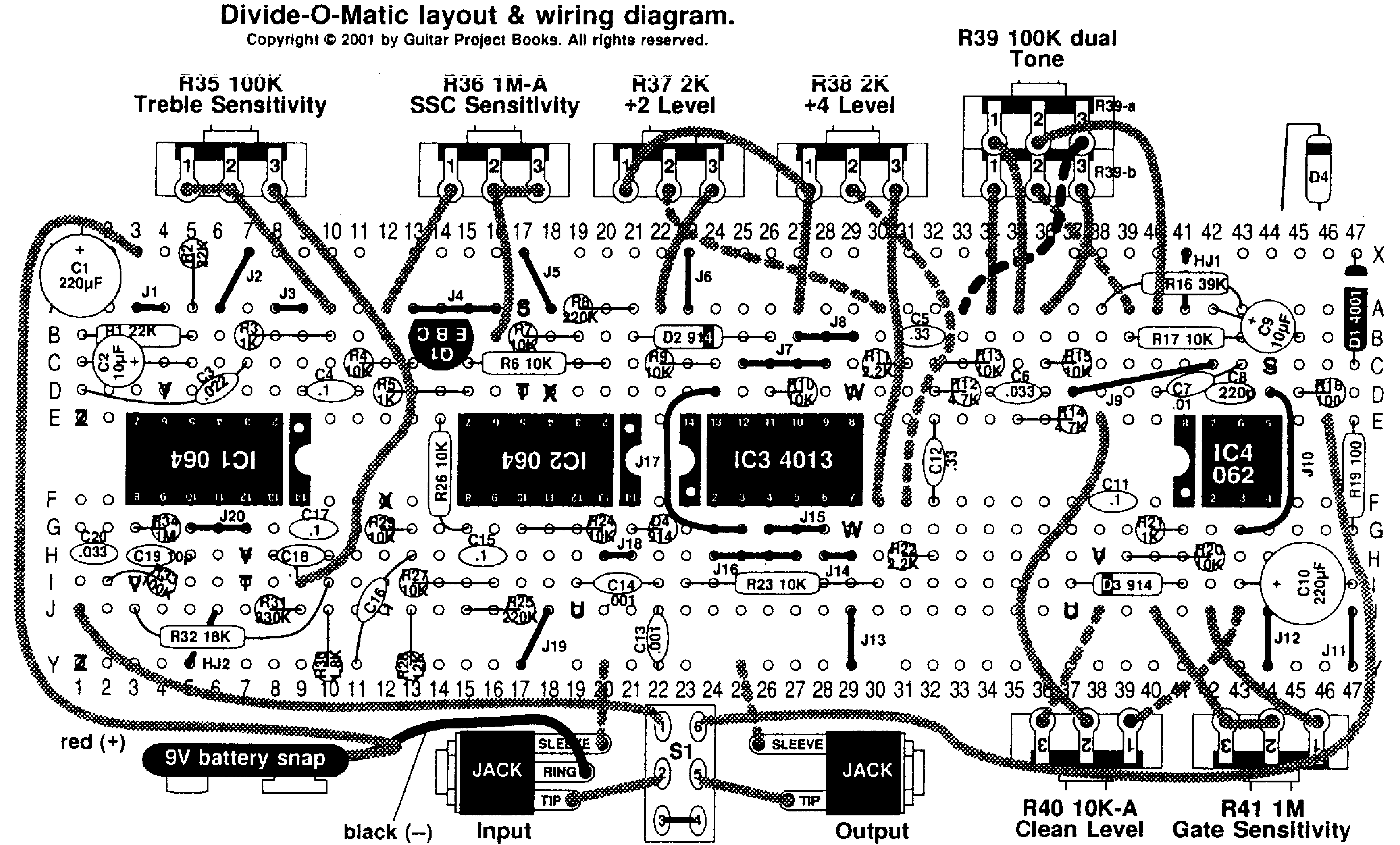
This box has two more panel controls than absolutely necessary—R35 and R36—but bringing these out lets the player tailor it to resist octave skipping with different axes and strings, over different regions of the fretboard.

The prototype drew 3.6 milliamps.

Divide-O-Matic schematic.



Divide-O-Matic layout & wiring diagram.



Parts List/Soldering Checklist

Circuit Board Radio Shack p/n 276-170; Hosfelt p/n 42-183; or DC Electronics p/n J4-404

IC Sockets

[ ] 14-pin for IC1; pin-1 goes in E9
[ ] 14-pin for IC2; pin-1 goes in E21
[ ] 14-pin for IC3; pin-1 goes in F23
[ ] 8-pin for IC4; pin-1 goes in F41

Resistors

[ ]	R1	22K	(red-red-org)	B1-B5
[ ]	R2	22K	(red-red-org)	X5-A5
[ ]	R3	1K	(brn-blk-red)	B7-B10
[ ]	R4	10K	(brn-blk-org)	C11-C13
[ ]	R5	1K	(brn-blk-red)	D12-D15
[ ]	R6	10K	(brn-blk-org)	C15-C20
[ ]	R7	10K	(brn-blk-org)	B17-B19
[ ]	R8	220K	(red-red-yel)	A19-A21
[ ]	R9	10K	(brn-blk-org)	C22-C24
[ ]	R10	10K	(brn-blk-org)	D26-D27
[ ]	R11	2.2K	(red-red-red)	C30-C31
[ ]	R12	4.7K	(yel-vio-red)	D32-D33
[ ]	R13	10K	(brn-blk-org)	C32-C34
[ ]	R14	4.7K	(yel-vio-red)	E35-E37
[ ]	R15	10K	(brn-blk-org)	C36-C37
[ ]	**[install hidden jumper HJ1 (X41-A41) before soldering R16]**			
[ ]	R16	39K	(org-wht-org)	A38-A43
[ ]	R17	10K	(brn-blk-org)	B39-B43
[ ]	R18	100	(brn-blk-brn)	D45-D46
[ ]	R19	100	(brn-blk-brn)	E47-G47
[ ]	R20	10K	(brn-blk-org)	H39-H42
[ ]	R21	1K	(brn-blk-red)	G40-G41
[ ]	R22	2.2K	(red-red-red)	H31-H32
[ ]	R23	10K	(brn-blk-org)	I23-I30
[ ]	R24	10K	(brn-blk-org)	G17-G20
[ ]	R25	220K	(red-red-yel)	J15-J17
[ ]	R26	10K	(brn-blk-org)	E14-G15
[ ]	R27	10K	(brn-blk-org)	I13-I16
[ ]	R28	12K	(brn-red-org)	J13-Y13
[ ]	R29	10K	(brn-blk-org)	G12-G13
[ ]	R30	1.8K	(brn-gry-red)	J10-Y10
[ ]	R31	330K	(org-org-yel)	J8-J9
[ ]	**[install hidden jumper HJ2 (J6-Y5) before soldering R32]**			
[ ]	R32	18K	(brn-gry-org)	J3-I10
[ ]	R33	100K	(brn-blk-yel)	I2-I4
[ ]	R34	1M	(brn-blk-grn)	G3-G4

Bare Wire Jumpers

[ ]	J1	A3-A4
[ ]	J2	X7-A6
[ ]	J3	A8-A9
[ ]	J4	A13-A16
[ ]	J5	X17-A18
[ ]	J6	X23-A23
[ ]	J7	C25-C28
[ ]	J8	B27-B29
[ ]	J9	D37-C42
[ ]	J10	D44-G43
[ ]	J11	J47-Y47
[ ]	J12	J44-Y44
[ ]	J13	J29-Y29
[ ]	J14	H28-H29
[ ]	J15	G26-G28
[ ]	J16	H24-H27
[ ]	J17	D24-G25
[ ]	J18	H20-H21
[ ]	J19	J18-Y17
[ ]	J20	G5-G7

Capacitors

[ ]	C1	220µF	X1-A1 (+ lead to X1)
[ ]	C2	10µF	C1-C5 (+ lead to C5)
[ ]	C3	0.022	D1-C7
[ ]	C4	0.1	D9-D11
[ ]	C5	0.33	B31-B32
[ ]	C6	0.033	D34-D36
[ ]	C7	0.01	D40-C43
[ ]	C8	220pF	D42-D43
[ ]	C9	10µF	A42-B45 (+ lead to A42)
[ ]	C10	220µF	I43-I47 (+ lead to I43)
[ ]	C11	0.1	F38-F39
[ ]	C12	0.33	E32-F32
[ ]	C13	0.001	J22-Y22
[ ]	C14	0.001	I19-I22
[ ]	C15	0.1	H15-H16
[ ]	C16	0.1	H13-Y11
[ ]	C17	0.1, 5%	G9-G10
[ ]	C18	0.1, 5%	H8-H10
[ ]	C19	10pF	H3-H4
[ ]	C20	0.033	H1-H2

Flying Jumpers (insulated wire)

[ ]	SS	A17-C44
[ ]	TT	I7-D17
[ ]	UU	J19-J37
[ ]	VV	I3-H38
[ ]	WW	D29-G29
[ ]	XX	F12-D18
[ ]	YY	D4-H7
[ ]	ZZ	E1-Y1

Semiconductors

[ ]	D1	1N4001	X47-C47 (banded end to X47)
[ ]	D2	1N914	B21-B26 (banded end to B26)
[ ]	D3	1N914	I37-I41 (banded end to I37)
[ ]	D4	1N914	G21-G22 (banded end to G21)
[ ]	IC1	TL064 quad op amp; pin-1 to E9	
[ ]	IC2	TL064 quad op amp; pin-1 to E21	
[ ]	IC3	4013; pin-1 to F23	
[ ]	IC4	TL062 dual op amp; pin-1 to F41	
[ ]	Q1	2N3906; emitter (E) to B15, base (B) to B14, collector (C) to B13	

Potentiometers (T=terminal)

[ ]	R35	100K	T1 to T2 and to A10, T3 to I9
[ ]	R36	1M-A	T1 to A12, T2 to T3 and to B16
[ ]	R37	2K	T1 to A27, T2 to A30, T3 to A22
[ ]	R38	2K	T1 to A27, T2 to F31, T3 to F30
	R39	100K dual	
[ ]	R39-a		T1 to A35, T2 to A40, T3 to A33
[ ]	R39-b		T1 to A34, T2 to A39, T3 to A36
[ ]	R40	10K-A	T1 to J43, T2 to E38, T3 to J38
[ ]	R41	1M	T1 to J42, T2 to T3 and to J40

Jacks (T=terminal)

[ ] input jack (1/4" 3-terminal/stereo): tip to T2 of S1; ring to negative (–) battery lead, sleeve to Y20
[ ] output jack (1/4" 2-terminal/mono): tip to T5 of S1, sleeve to Y25

Switches (T=terminal)

[ ] S1 (DPDT stomp switch): T1 to J1, T2 to tip of input jack, T3 to T4, T5 to tip of output jack, T6 to E46

9V Battery Leads

[ ] black (–) lead to ring of input jack
[ ] red (+) lead to X3

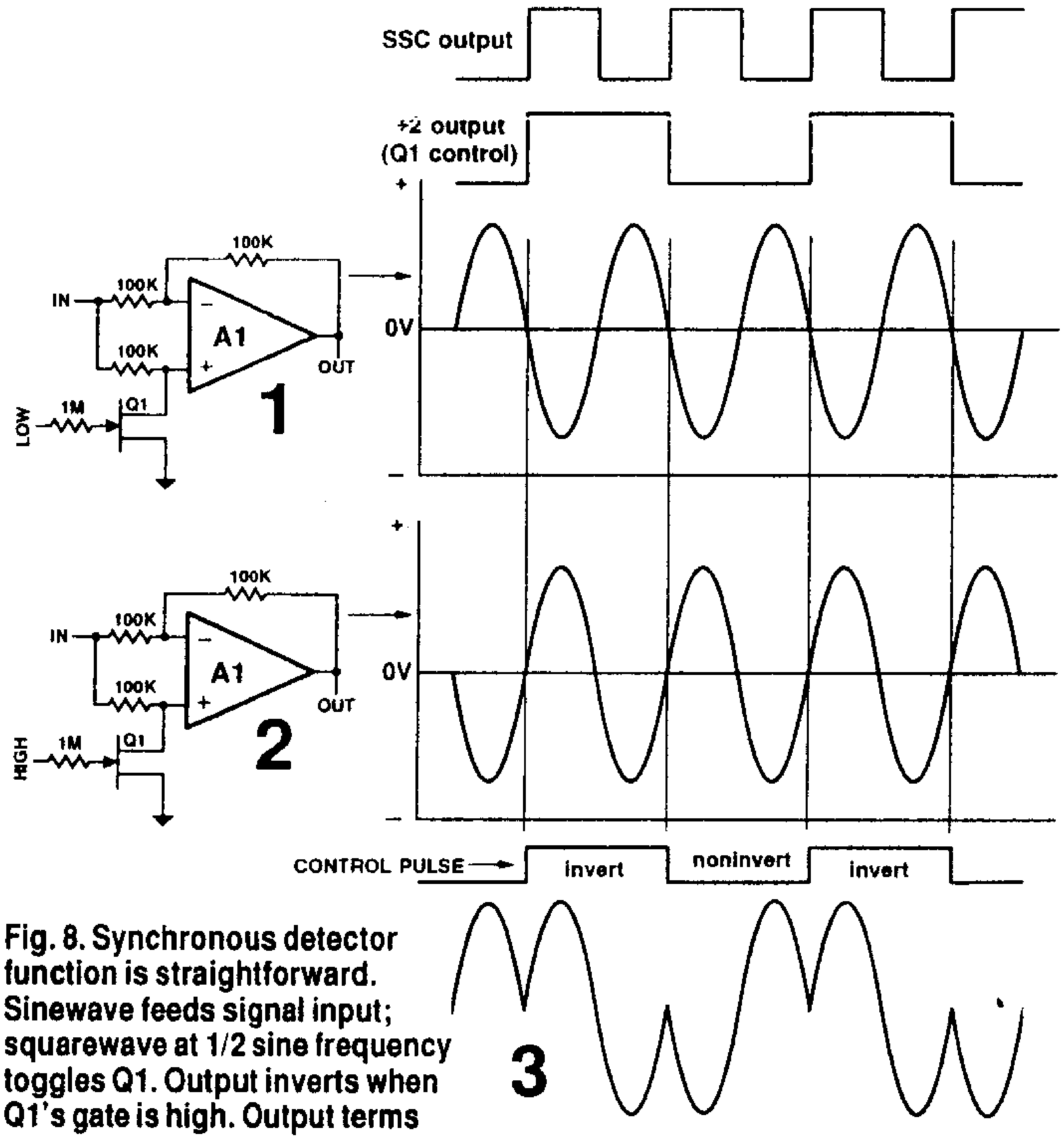


Fig. 8. Synchronous detector function is straightforward. Sinewave feeds signal input; squarewave at 1/2 sine frequency toggles Q1. Output inverts when Q1's gate is high. Output terms include 1/2 and 3/2 the sinewave frequency. Squarewave transitions coincide with sinewave zero-crossings. Inversion of either raw SSC output or ÷2 feed does not affect circuit's operation.

(continued from page 3)

can optimize a filter for the treble region.

Q. What tells me that I have an adequate filter?

A. The SSC should track the fundamental for two full seconds on over 90% of notes, from the open low-E up to the 12th fret on the B-string.

Q. What about the other 10%?

A. On dead frets, the string vibrates at the fundamental for only a fraction of a second. The SSC has nothing to key on except harmonics. Weak frets show enough fundamental to trigger SSC, but skip early in the note's decay. On dead and weak frets, move up the neck and play the notes on a lower string.

Q. Say I've got the SSC up over 90% without octave skipping. What next?

A. Couple the SSC output to two sequential ÷2 blocks to give first and second sub-octaves.

Q. Such as the 4013 circuit in Fig. 1?

A. Right.

Q. What's the purpose of D1-R4 in that circuit?

A. To ensure triggering of the 4013 from the output of an 06X op amp.

Q. At this point in design I have raw ÷2 and ÷4 feeds. The block diagram shows them entering a gate or expander. Which should I use, and why?

A. That depends on your design goals. Working up Divide-O-Matic, we wanted the note to cut off promptly when we muted the string. That dictated a gate. The circuit (IC4-a, IC2-d/-2, Q1) closely resembles the gates used to mute the carrier oscillator in ring modulators [*see Vol. 8, No. 1—Ed.*].

Q. What about expansion?

A. Ideally, you'd impress the instrument's dynamics on the constant-amplitude squarewave train; what we call dynamic restoration. This is too big

Fig. 9. Operation of this indirect divider becomes clear by treating it as two separate circuits. First circuit consists of A1, Q1, and all resistors. 1—Circuit multiplies input by 0.5 when Q1 is pinched off. 2—Circuit multiplies input by −0.5 when Q1 is turned on. Second circuit consist of C1 and D1, which form a positive peak-to-peak detector, sometimes called a DC restorer (see Fig. 10). In this case, C1-D1 shift the AC input signal almost entirely above ground. 3—By toggling Q1 at half the sinewave frequency, output of A1 resembles a sinewave of half the input frequency. 4—In practice, the two output halves do not align perfectly. Trimming the DC bias present at D1's anode achieves the nearly perfect alignment implied in #3. 5—Circuit demands specific timing arrangements. If raw SSC output is inverted, A1's output resembles #5, which is musically useless. If squarewave transitions do not coincide with sine peaks, waveform shape falls somewhere between #3 and #5. If waveform #5 is observed when working up an effect, reverse the orientation of D1, or place an inverting buffer between the SSC and the ÷2 circuit, and the waveform will normalize.

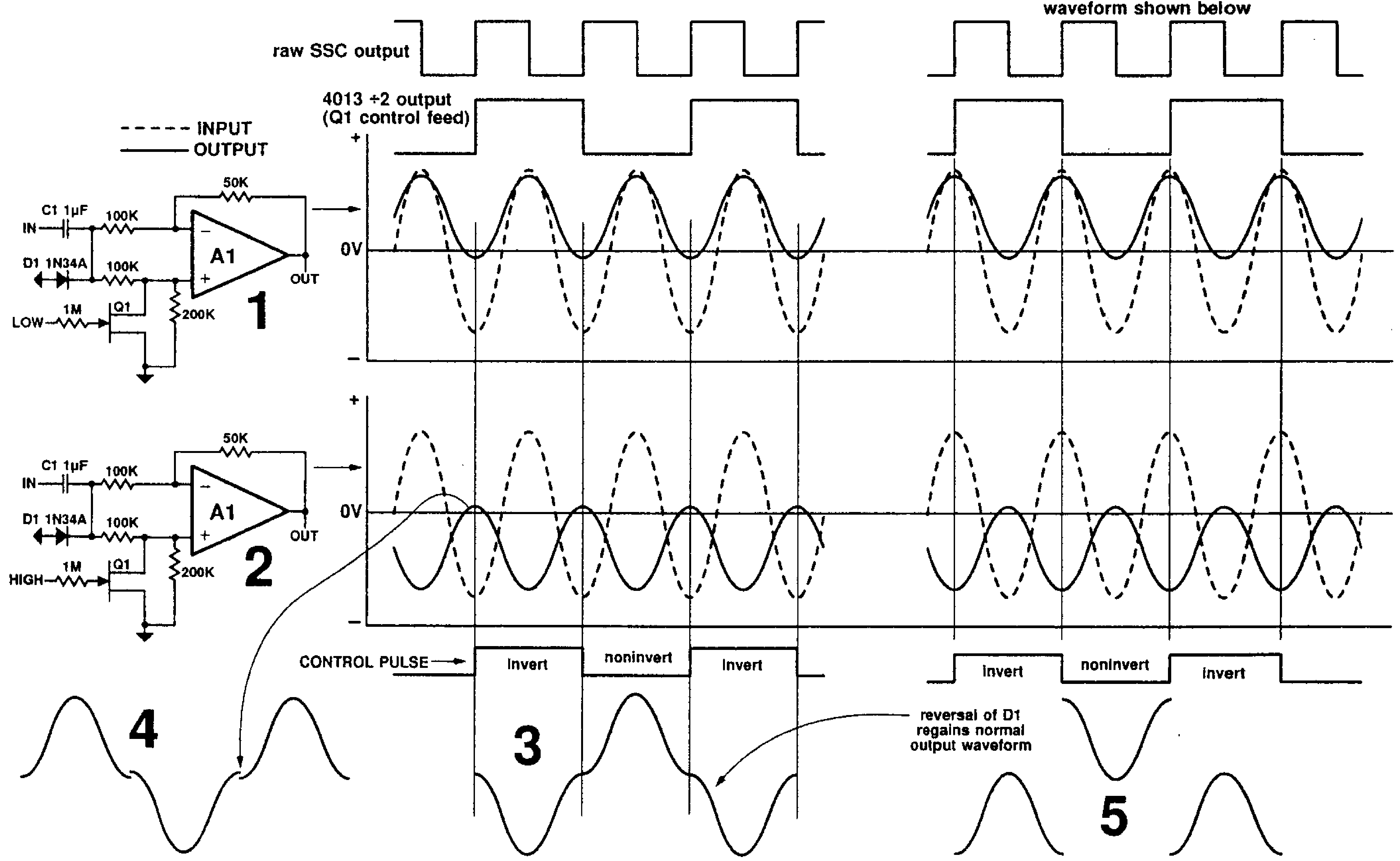
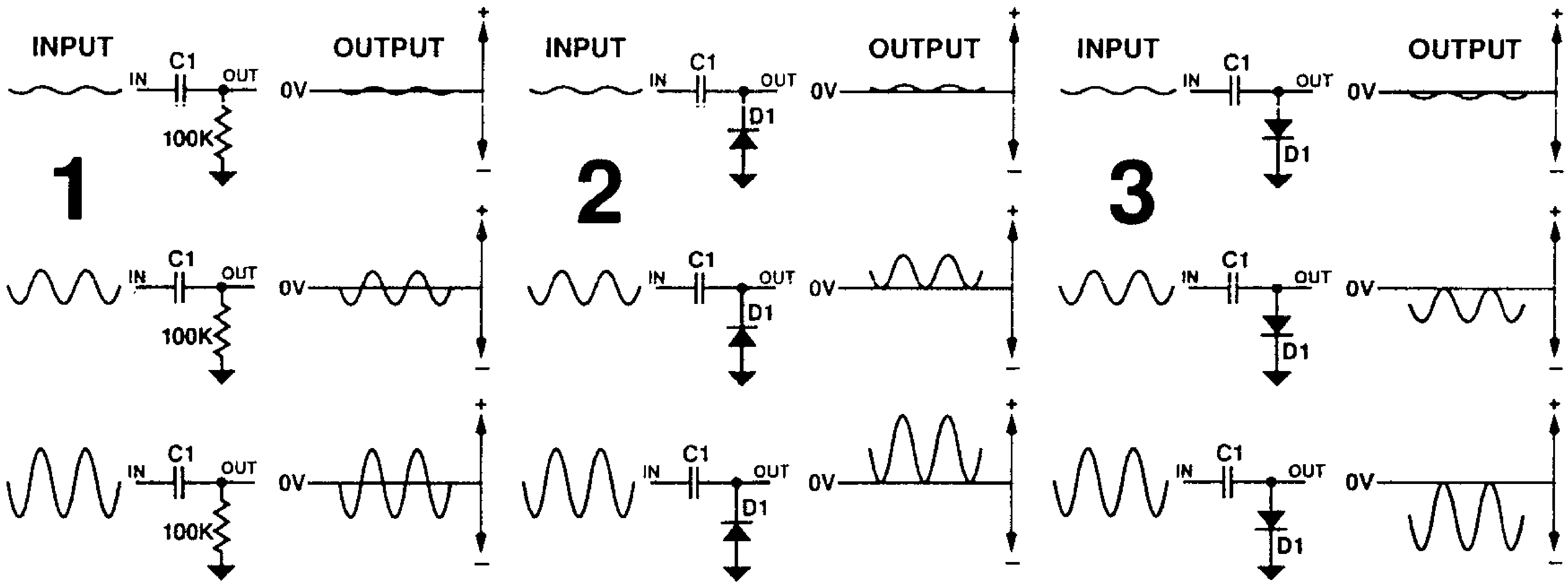


Fig. 10 A closer look at the action of the DC restorer (C1-D1) in Fig. 9. 1—Audio voltage negotiates capacitor pretty much unchanged, so long as cap exhibits low reactance at input frequency. 2—Replacing R1 with a germanium diode prevents output from dipping below ground (a slight dip occurs, due to D1's forward drop). 3—Reversal of diode polarity creates a negative output.



a topic to cover here in detail. Fig. 15 shows one approach that's simple and effective.

Q. What's the best output filter for a direct-divider box?

A. Use the filter that sounds best to you. Lowpass smooths the sound, but never really loses the squarewave buzz. Treble boost is useful in a distortion box; bass boost helps if your amp will reproduce the synthetic lower octaves.

Q. Let's say I want to work up an indirect divider for guitar.

A. Perfectly feasible—in theory. But in our opinion you won't find the synchronous detector satisfactory for wideband duty in guitar. Its 3/2 term sounds unmusical, and is impossible to remove, over a wide frequency range, using a fixed filter. It makes a good choice for subharmonic synthesis confined to the guitar's bottom octave. The narrow range lets you filter sharply enough to remove the 3/2 term.

The search for some means to isolate the subharmonic over a wide frequency range brought us to self-tuning filters. They work, but bring their own problems and limitations.

Q. For example?

A. Check out Fig. 13. The self-tuning filter's basic concept is found in Ref. 3. Our embodiment uses precision diodes, and substitutes cheap transconductance amps for the costly analog multipliers specified in the patent, which were further limited in that their control voltage had to be kept above ground. OTAs respond over the full output swing of the 06X op amp in a ±15V system.

The filter works best when fed a signal of at least 10V peak. The lowpass output gives the smoothest waveform. Distortion falls as filter Q rises, but tuning time rises with filter Q. This circuit can convert a squarewave to a low-distortion sinewave, but doing so requires a high Q that prolongs tuning and bumps the filter into chaotic modes if the input frequency changes quickly.

Q. Is this tracking filter the fastest available?

A. Ref. 5 describes a tracking filter that tunes within one cycle, but uses a digital-to-analog converter to generate the tuning voltage.

Q. Other approaches?

A. Fig. 14 shows a non-feedback tracking filter, based on the LM2917 frequency-to-voltage converter.

Q. Getting back to indirect-divider stompboxes, what about one based on the circuit in Fig. 9?

A. Ref. 7 is the schematic of a commercial product that takes that approach.

Q. I have questions about the projects.

A. Fire away.

Q. In Sub-O-Matic, what's going on with the input bandpass filter, IC2-b/-c/-d?

A. The approach is called *cascaded pole synthesis*, straight out of Ref. 6. Referring to Sub-O-Matic's schematic, the first and third filters, IC2-b and -d, have a Q of ~6; the second filter, IC2-c, has a Q of ~3. We tweaked R36 to give the passband a slight downward slope like that of Fig. 5.

Q. In Sub-O-Matic's post-filter, (IC4-c), the ratio of C8 and C7 seems a radical departure from values recommended in Ref. 6.

A. We found these values while viewing input and output on the scope, while changing filter caps. This filter gives a much more sinusoidal output than conventional lowpass filters up to 4th order with 3-dB dips. It also gives a peak over the range 25–35 Hz that helps compensate for falling frequency response in most speaker systems.

Q. Why no raw subharmonic output on Sub-O-Matic?

A. Tight board. If you want the raw sub output, solder a 100-ohm resistor in series with a 10µF nonpolar cap to the wiper of R45; tie the cap's free end to the tip of an output jack; tie the jack's sleeve to circuit ground.

Q. Can I double those components to get two outputs for subwoofers that need them?

A. Yes.

Q. Regarding Sub-O-Matic II: (1) Why use a trimpot for D3? (2) Why is D3

Fig. 11. Sinewaves 1 and 2 are identical but for a phase separation of 90 degrees. This relation is called quadrature, and is useful, among other reasons, for the fact that zero-crossing transitions for wave 1 coincide with peaks for wave 2, and vice versa.

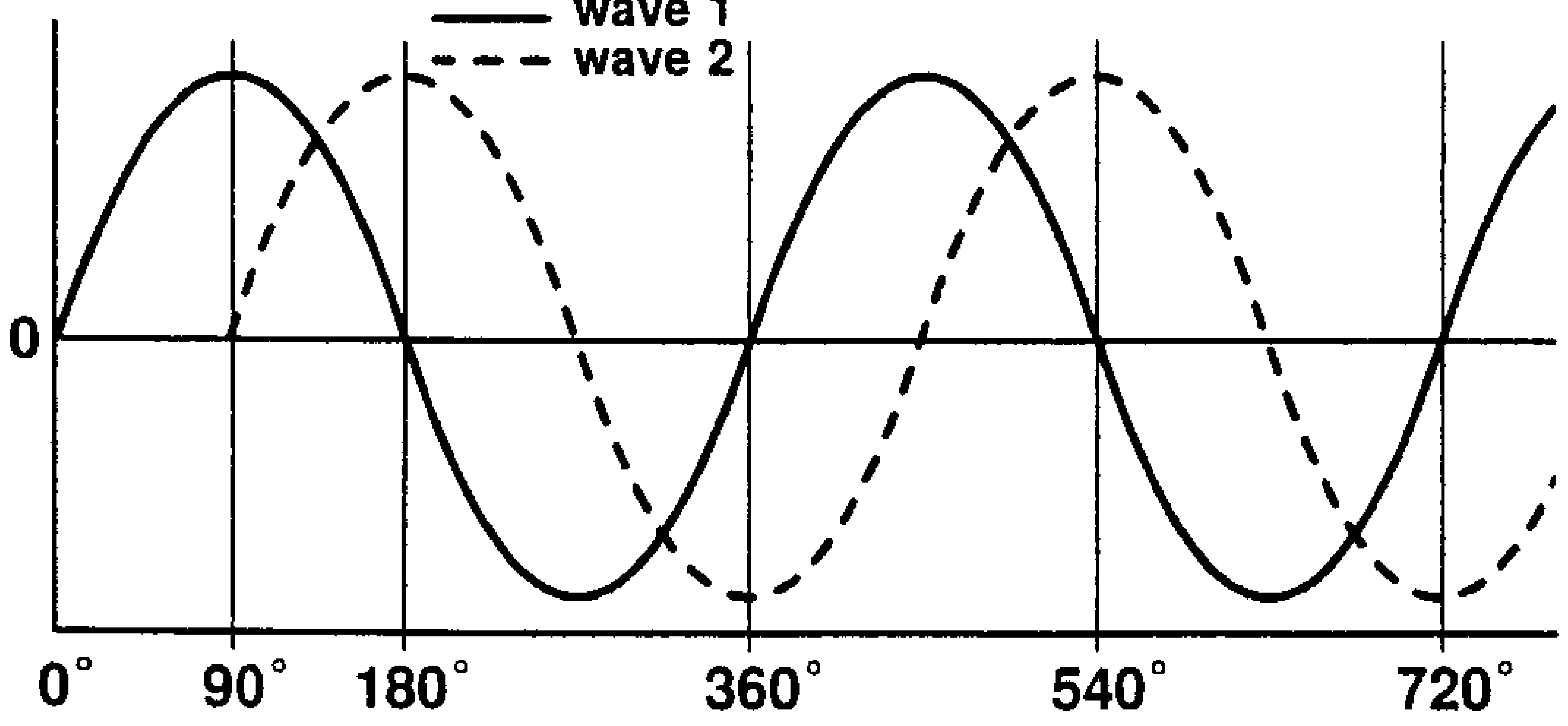
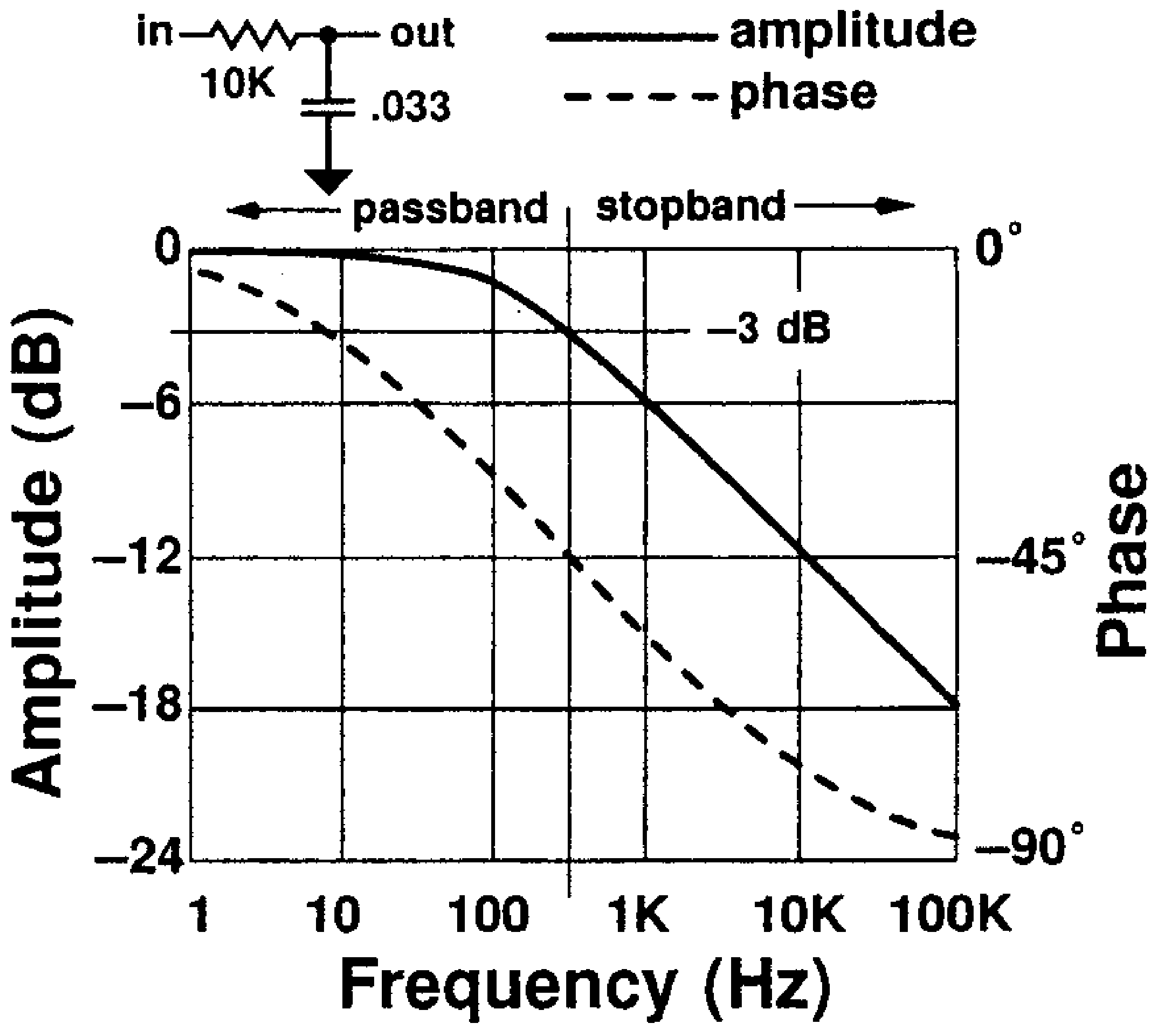


Fig. 12. Graphs approximate the phase and amplitude changes in a first-order, passive lowpass filter that corners at ~500 Hz.



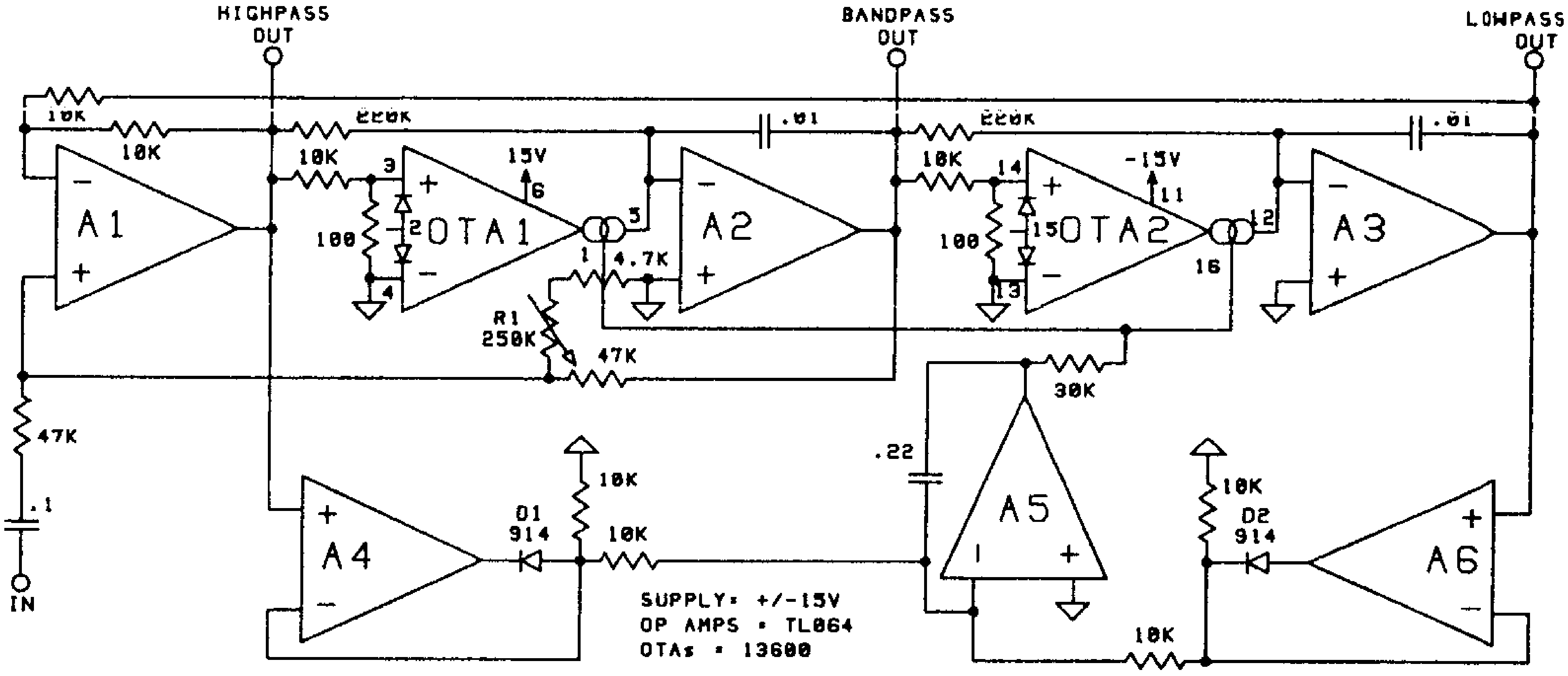


Fig. 13. Self-tuning filter based on principles outlined in Ref. 3. State-variable filter is made tunable by replacement of tuning resistors with LM13600 OTAs. Precision diodes compare amplitudes at highpass and lowpass outputs, which are equal when the filter is tuned to the input frequency. Integrator A5 generates an error voltage that tunes filter. These component values let filter track over ~200–2000 Hz. Filter converts squarewave input to sinewave output whose distortion falls as Q rises (achieved by reducing R1). Filter is sensitive to input amplitude, responding faster to larger signals. Response time rises with Q.

upside down compared to D1 in Fig. 9?

A. Referring to Sub-O-Matic II's schematic, the optimum bias for D3 is not 1/2V+. Trimming is required to optimize the point at which the two half-waves combine. Fig. 9–4 shows the type of waveform that results without the trim.

D3's polarity is reversed compared to D1 in Fig. 9, for reasons implicit in the timing relationships shown in Fig. 9. If D3 were oriented the same as D1 in Fig. 9, the output of IC4-b would look like Fig. 9–5. The timing relationship between the input waveform and the SSC output dictates the orientation of D1 (D3 in Sub-O-Matic II).

Q. I took the liberty of obtaining Refs. 1 & 2. Both specify gating circuitry for the subharmonic generator. Why didn't you gate the three Sub-O-Matics?

A. (1) Lack of room on the circuit board. (2) More importantly, listening tests showed that these projects didn't need gating. If we were designing commercial products that have to keep their cool under difficult conditions, yes, we'd add a gate, keyed to the bass content of the input program.

Q. I'm not entirely clear on the subharmonic generator shown in Figs. 4–2 & 9, which you've used in Sub-O-Matic II.

A. This circuit is actually two separate circuits. Referring to Fig. 9, the first subcircuit consists of A1, Q1, and all resistors. It acts the same as the synchronous detector, except that toggling Q1 multiplies the input alternately by 0.5 and –0.5.

Q. That leaves C1 and D1 to make the other circuit.

A. Believe it or not, those two components form a peak-to-peak detector.

Q. What's that?

A. A peak-to-peak detector is a form of AC-to-DC converter, sometimes called a *DC restorer.* It prevents the negative audio swing from dipping below ground (Fig. 10); or, if D1 is reversed, keeps the positive audio swing from rising above ground.

Q. As I understand it, comparator transitions must coincide with zero-crossing in the synchronous detector, and must coincide with sinewave peaks in the other indirect divider. How do you achieve those timing relationships?

A. Let's take zero-crossing sync first. This turns out to be easy, because several common comparator circuits automatically switch at the sinewave's zero-crossing point. In fact, such comparators are often called zero-crossing detectors (ZCDs). Given the right resistor/capacitor values, Figs. 2–1 and 2–2 act as ZCDs; specifically, in Fig. 2–2, use 10K resistors and a 10µF cap.

As for the indirect divider shown in Fig. 9, several approaches sync comparator transitions to sinewave peaks. One precedes a ZCD with a quadrature phase shift network.

Q. What does that mean?

A. Two sinewaves separated by exactly 90 degrees are said to be *in quadrature*; *quadrature phase shift* means *90 degrees of phase shift.* You can see from Fig. 11 that a sinewave's zero-crossing point is always 90 degrees away from a peak. To make a peak-crossing detector, first feed the signal through a quadrature phase shift network, then to the ZCD. Relative to the original sine input, squarewave transitions then coincide with sinewave peaks.

Q. So, in Sub-O-Matic II, you're saying that R30-C3 forms a quadrature phase shift network? Looks like a passive lowpass filter feeding an amp (IC1-a) with gain of ~21.

A. Correct. But filters shift phase as well as amplitude (Fig. 12). In this case, the stopband phase-shift approaches –90 degrees.

Q. What do you mean, "stopband phase shift"?

A. R30-C3 form a lowpass filter that passes tones below ~1.6 Hz. The tones we're using live well above 20 Hz. So, those frequencies are in the filter's *stopband.* Tones below 1.6Hz occupy the *passband.* The filter's phase shift affects mainly stopband frequencies.

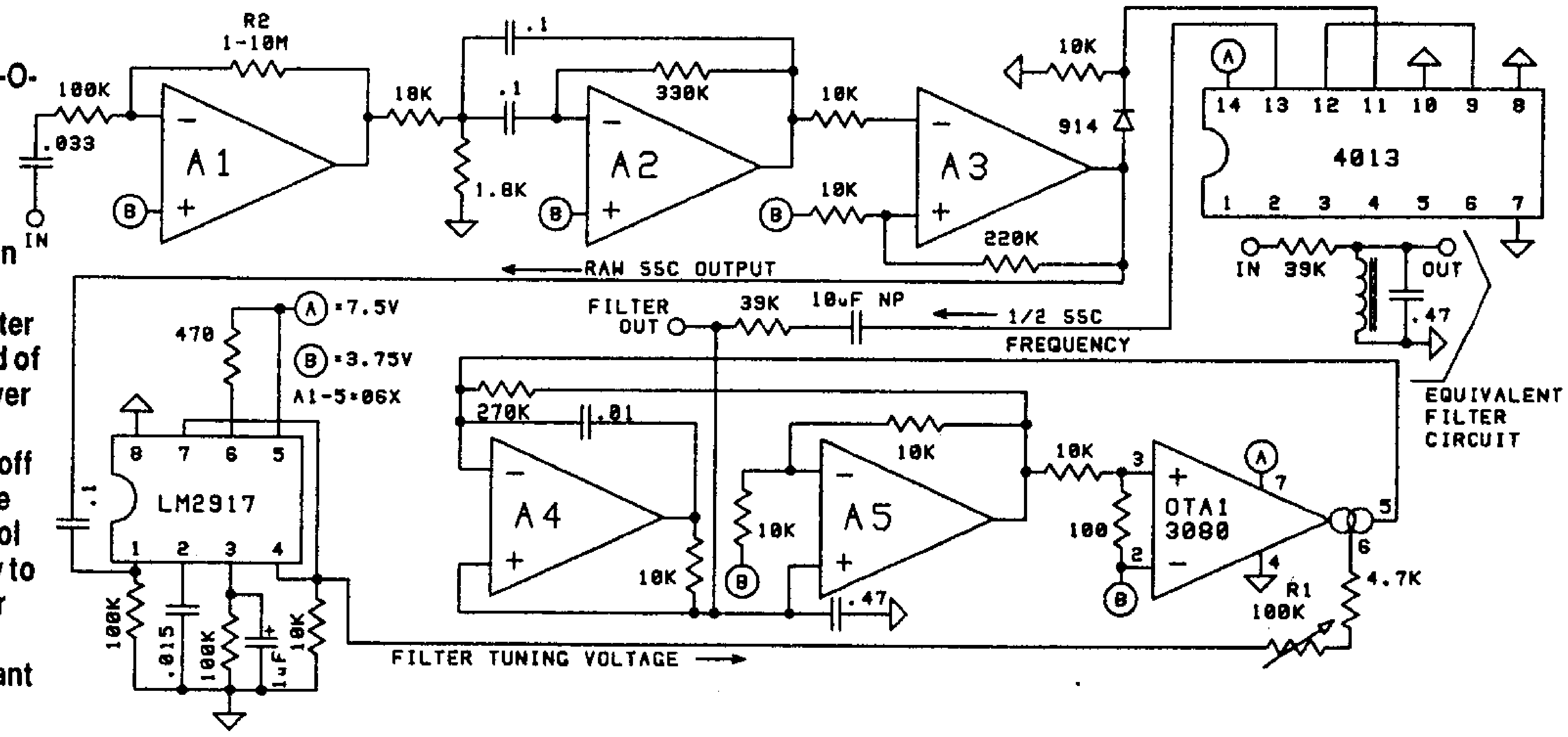


Fig. 14. Non-feedback self-tuning filter uses input chain similar to that of Divide-O-Matic (A1, A2, A3, 4013), to convert sine input to squarewaves; 4013 halves the squarewave train. The raw squarewave output feeds an LM2917 frequency-to-voltage converter; output voltage is taken at pin-4 and is proportional to input frequency. This voltage controls the center frequency of a bandpass filter comprised of A4, A5, C1, and OTA1. This filter tunes over the approximate range 40–150 Hz. The input to the filter is the ÷2 signal coming off the 4013. The undivided squarewave tone is used to drive the 2917, to reduce control voltage ripple. Trimming R1 is necessary to get the filter to track reasonably well over the input range 40–150 Hz. Filter output approximates a sinewave, but has constant amplitude.

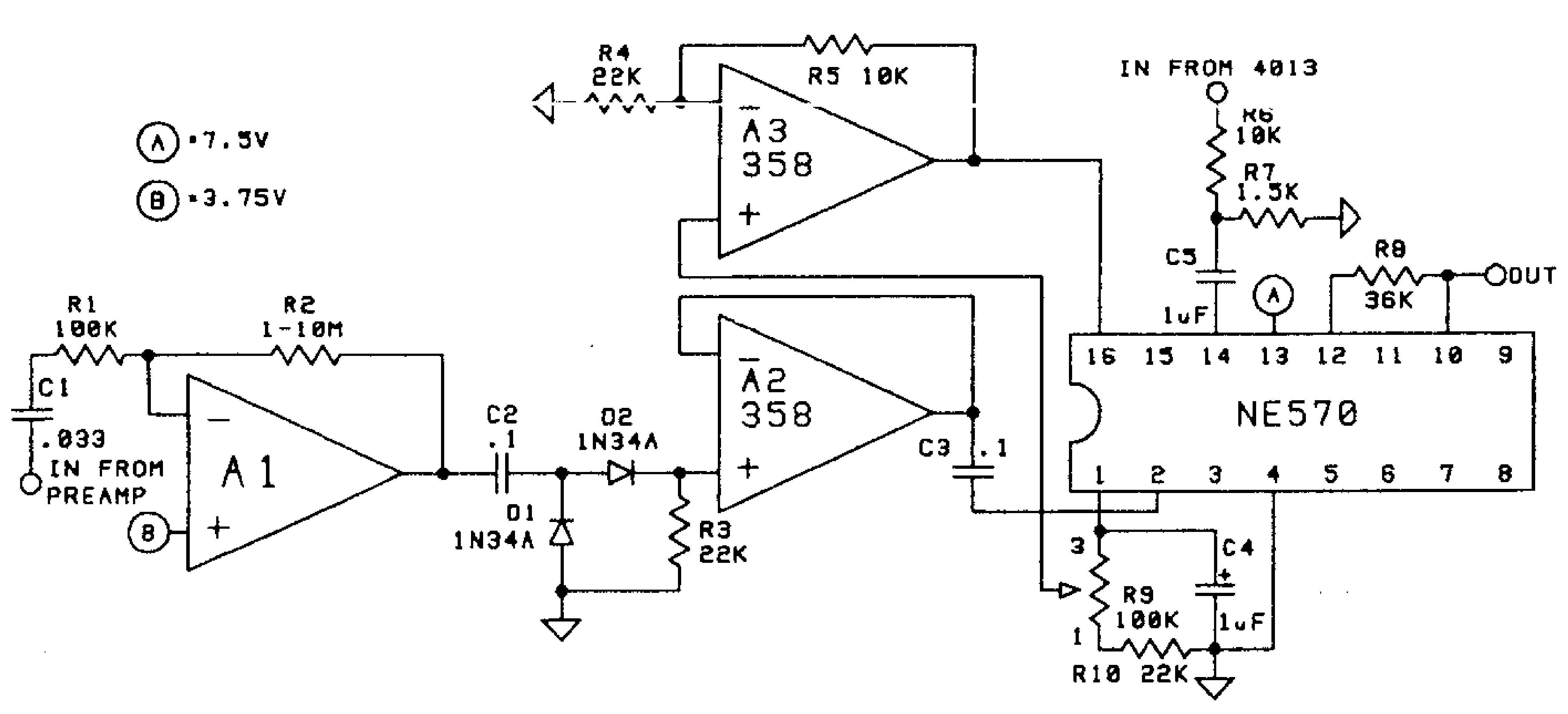


Fig. 15. "Dynamic restorer" suitable for use in divider box. Raw axe feed, or preamp output, couples through C1 to A1, an amp whose gain is fixed between 10 and 100 by value of R2; A1 can be any op amp. A1's output couples to a DC restorer consisting of C2 and D1, feeding D2-R3, which makes a 0.5V dead zone for the audio feed; buffered by A2, whose output feeds the rectifier input (pin-2) of one channel of an NE570. Control voltage is taken at pin-1, coupling to pot R9, which varies the degree of upward expansion, thence to buffer amp A3 (A2 and A3 must be 358, 324, TS272, or other type whose input range includes the negative supply). A3's output couples to pin-16, the control port of a VCA formed by the 570's other channel. The 4013 feeds the VCA's input; the 7.5V peak amplitude is reduced to about 1V by divider R6-R7. This circuit responds smartly to playing dynamics, yet exhibits modest sustain and automatic gating.

Q. Why not use an allpass filter, to avoid amplitude loss?
A. An allpass filter has a sharper phase slope than a first-order lowpass filter's. The gentler the slope, the wider than band of frequencies for which phase shift approximates 90 degrees. When we tested allpass filters while working up Sub-O-Matic II, comparator transition points wandered away from peak as we varied the frequency from 40 Hz to 100 Hz. The lowpass filter kept the transitions very close to peak over that range.

Other, simpler approaches keep a shallow phase slope without losing amplitude. For example, "magic" component values cause the comparator shown in Fig. 2–2 to cross at a sine input's peak, over the range ~80–400 Hz (make both resistors 4.7K, make the cap 0.022µF; DC couple to the prior stage, which should be biased at 1/2V+ in a single-supply system, at ground in a dual-supply system). The circuit requires a clean sine input to maintain the desired phase relationship.

Like zero crossing, quadrature is a key concept in electronics, out of the many instances that call for a 90-degree offset between events. Subharmonic synthesis gives another example.

Q. Could you explain Divide-O-Matic's tone control?
A. The circuit is an active bass & treble control, straight out of op-amp databooks, and explained in some detail in the *Cookbook* (p. 119). All we've done is wire a dual pot (R39) such that CCW rotation boosts bass and cuts treble; CW rotation does the reverse.

Q. Could I split these into independent bass & treble controls?
A. Sure.

References

1. US Patent No. 4,182,930: Detection and Monitoring Device

2. US Patent No. 4,698,842: Audio Processing System for Restoring Bass Frequencies

3. US Patent No. 3,978,420: Self-Tuning Filter

4. US Patent No. 3,714,588: Self-Tuning Filter

5. US Patent No. 3,978,416: Filter which Tracks Changing Frequency of Input Signal

6. Lancaster, Don: *Active-Filter Cookbook*, Howard W. Sams, 1975, ISBN 0-672-21168-8

7. Schematic of Boss® "OC-2" stompbox, viewed over the internet

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