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3.30	\$9.10	25.0	\$34.00
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4.70	\$10.15	33.0	\$45.60
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6.20	\$12.20	47.0	\$62.80
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2.20	\$3.10	20.0	\$13.45
3.30	\$3.70	22.0	\$14.35
3.90	\$4.10	25.0	\$15.90
4.70	\$4.40	30.0	\$17.30
5.00	\$4.60	39.0	\$21.00
5.60	\$5.05	47.0	\$25.15
6.00	\$5.25	50.0	\$28.05
6.80	\$5.70	68.0	\$34.70
8.20	\$6.15	82.0	\$41.90
9.00	\$6.40	100.0	\$52.25
10.0	\$6.70		

A Multimode Filter/Resonator for the Sound Strobe and Other Audio Applications

This highpass/bandpass/lowpass filter has a tunable f_o (20Hz-20kHz) and a wide Q range (0.5-100) for audio filtering, resonant tone pulse generation, auditory music analysis, measurements, and more.

The Sound Strobe (*aX* March '06, and reviewed by Ed Simon in *aX* April '07) generates six selectable-spectrum (but wide-band) pulses that are very revealing of speaker transient precision and in-room sound clarity. The tunable, variable resonance filter I describe here allows focusing the pulses' spectrum into a bandwidth (BW) ranging from 2.5 octaves to less than 1/48 octave (1/4 semi-tone). Then, with octave switching or a 20Hz – 20kHz sweep control of f_o , you can hear transient response, frequency response smoothness, image focus, and room acoustics effects, as a function of frequency. (f_o is the bandpass output's center frequency.)

With high Q (resonance) settings, the Sound Strobe pulses produce, at the bandpass (BP) output, an exponentially-decaying sinusoidal tone burst, at the filter's f_o . These start with a clean and precise onset, producing a sound that resembles musical tone transients. Because of this, you hear the characteristics of speakers and room colorations with recognizable correlation to musical reproduction quality.

"AUTO Q" MODE

In addition to a manual mode (where you can adjust Q from 0.5 to 30), there's an "Auto" mode. Here, the Q increases with the f_o setting, ranging from 7.3 at

20Hz to over 100 at 20kHz. Used with the Sound Strobe's "25ms exponential" pulse (a full-band step pulse rolled off below 6Hz), the filter's BP output has sufficient resonant decay time to hear a distinct tonality, while being short enough to hear transient-response details. As you tune f_o , the pulse sound has these characteristics:

1. f_o range of 20-50Hz: resembles bass drums of various sizes.
2. 40-300Hz: the fundamental tones of a plucked string bass.
3. 300Hz-1kHz: chimes and bell transients, somewhat guitar-like.
4. 1kHz-3kHz: the metallic transients of small bells, xylophones.
5. Above 3kHz: high-pitched transients such as small triangles.

As you tune f_o over the full audio band, a multitude of speaker and room anomalies can be revealed. Also, slowly sweeping f_o allows you to hear the sound system's coherence and smoothness (or lack thereof) of frequency response. And, as with the raw Sound Strobe pulses, you can easily discriminate the direct speaker sound from room effects. But with this filter, you can "hone in" on any specific frequency.

A switchable +1.82dB/octave slope equalizer (EQ), when used with the Sound Strobe 25ms exponential pulse and the filter's "Auto" Q mode, main-

tains a constant energy per pulse regardless of f_0 .

These precise-transient exponential tone bursts, as their f_0 is swept, will reveal many details regarding overall speaker sound clarity, timing precision, and spatial image focus.

OTHER FILTER USES

1. Auditory spectral analysis: Playing music through the filter (BP output) can be informative in several ways. With the Q (manual mode) set to about 4, the -3dB BW is about 1/3 octave (4.31 is the exact Q value). This is useful to explore the frequency range of various musical sources; with higher Q settings you can select individual instrument tone harmonics.

But perhaps the greatest use is in identifying the frequency ranges where there's some coloration or distortion that you've previously noticed. For this purpose, it's useful to feed one of the stereo channels through the filter, so you can tune the f_0 while also hearing the full-band sound from the other channel.

2. You can use the highpass (HP) and lowpass (LP) outputs with music to hear the effects of bandwidth restriction, and to determine the bass depth and HF extension in a particular piece of music. You should set the Q to 0.7, because this produces a Butterworth rolloff (maximally flat without peaking) of 12dB/octave.

3. Noise measurements: The HP and LP filtering can be useful in determining the noise voltage above or below some chosen frequency. The HP output is also useful to attenuate AC line hum.

4. Notch filter: A notch (about 40dB deep) response is easily generated by summing the LP and HP outputs with two 1k resistors. A low Q produces a wide notch, and higher Q values make the notch narrower. The passband voltage gain will be 0.5.

5. Any general-purpose applications where tunable filtering is needed.

6. Spectrum analysis: You can use the BP output with an AC voltmeter to measure audio waveform levels in tunable frequency bands.

7. Crossover design: Using the LP and

HP outputs, with one inverted and Q set to 0.5, lets you experiment with CO frequency with a tunable second-order CO.

STATE VARIABLE FILTER BACKGROUND

In 1971, while working at ARP Instruments (where I designed the ARP 2600 synthesizer, made known by Stevie Wonder), I presented a paper at the Audio Engineering Society on "The Electrical Design and Musical Applications of a Voltage-Controlled Filter/Resonator." The circuit I designed became the ARP Module 1047 Multimode Filter/Resonator (**Photo 1**). This made the variable tonality effect in the rhythmic background of The Who song entitled "Won't Get Fooled Again."

Fast-forward 36 years: **Photos 2 and 3** show the prototype of the circuit described in this article. I haven't designed it into an enclosure yet; I'm waiting to see whether there's enough interest. See **Fig. 12** for a recommended control component layout.

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PHOTO 1: ARP Module 1047 (from 1971).

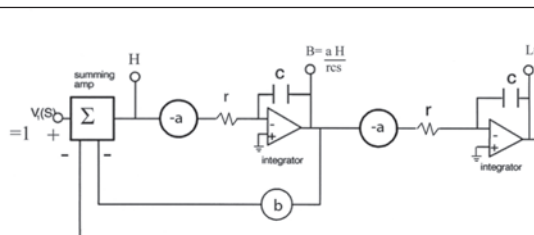


FIGURE 1: Laplace analysis of state variable filter.

$$B = \frac{aH}{rcs} \cdot L = \frac{a^2H}{r^2c^2s^2} \cdot H = 1 - \frac{abH}{rcs} - \frac{a^2H}{r^2c^2s^2}, H - 1 + \frac{abH}{rcs} + \frac{a^2H}{r^2c^2s^2} = 0 \quad S^2r^2c^2H + SabrcH + a^2H = S^2r^2c^2.$$

$$H = \frac{S \frac{a}{rc}}{s^2 + s \frac{ab}{rc} + \left(\frac{a}{rc}\right)^2}$$

$$\text{Highpass response } H = \frac{s^2}{s^2 + s \frac{ab}{rc} + \left(\frac{a}{rc}\right)^2}$$

$$\text{Bandpass response } B = \frac{S^2r^2c^2}{S^2r^2c^2 + Sabrc + a^2}$$

$$\text{Lowpass response } L = \frac{\left(\frac{a}{rc}\right)^2}{s^2 + s \frac{ab}{rc} + \left(\frac{a}{rc}\right)^2} \cdot fo = \frac{a}{2\pi rc}.$$

$$Q = \frac{1}{b}.$$

$$@fo = fo, H = B = L = Q.$$

TOPOLOGY, FREQUENCY, AND SQUARE WAVE RESPONSES

The topology and Laplace (frequency domain) analysis of the state-variable filter (also known as the "Biquad") are shown in Fig. 1.

Figure 2 shows the frequency responses on dB versus log frequency scales. The principal advantages of this topology are (1) you can obtain a very high Q (up to 1000) with stability, (2) the latter doesn't depend upon precise component value matching, and (3) you can achieve a very wide fo sweep range (over 1000:1) while maintaining response shapes and unity-gain passband regions.

In Fig. 1, the elements labeled "a" and "b" can be voltage controlled attenuators (VCAs) as in the design. The "a" elements must track, but can be either both inverting or both non-inverting. Inverting "a" elements provide non-inverting BP and LP outputs. Using a dual

log pot for the "a" elements can provide about a 100:1 fo sweep, and both "a" and "b" can be fixed resistors for fixed fo and Q applications.

Figure 3 shows all three outputs' responses to a 1kHz square wave, with Q values of 0.5 (non-resonant, left column) and 1.0 (slightly resonant, right column). Filter fo is set to 5kHz. The LP output is DC coupled, and at fo all three outputs have a voltage gain that's numerically (not dB) equal to Q.

FILTER RESPONSES TO SOUND STROBE "PINK PULSE"

The "pink pulse" (Fig. 4) is so-called because, like pink noise, its spectrum is flat on a per-octave, per third octave (and so on) basis. The filter's BP response, for a given Q, also maintains a con-

stant number of octaves (or fraction thereof) bandwidth regardless of f_0 . Therefore, the BP output from a pink pulse will maintain a constant energy per pulse as f_0 is changed. But with a constant Q setting, the pulse decay time is inversely proportional to f_0 .

To maintain constant pulse energy versus f_0 , the peak output voltage will increase proportionally to the square root of f_0 . With a step pulse filter input, the peak output voltage is constant with f_0 , so the decreasing decay time versus f_0 causes the pulse energy to decrease, inversely proportional to $\sqrt{f_0}$; that is, at -3dB/octave. But this is expected, because the spectrum of a step pulse, while downsloping at -6dB/octave on a per-Hz analyzed basis, has a -3dB/octave downslope on a per-octave basis. The latter, as

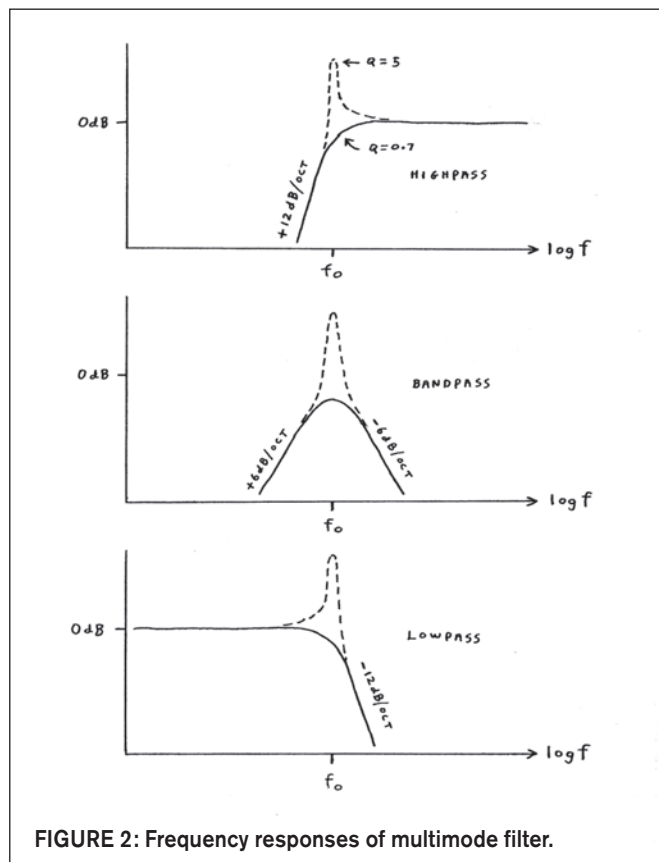
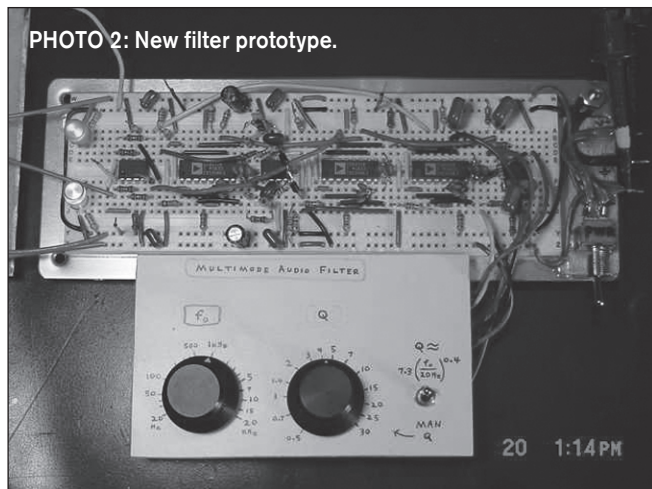


FIGURE 2: Frequency responses of multimode filter.

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explained in the Sound Strobe article, is the way we hear: we analyze spectra on a per-octave (or fraction) basis.

SOUND STROBE PULSE WITH HIGH-Q RESONANCE

In **Fig. 4**, the Q is a high value of 59, with $f_0 = 5\text{kHz}$. Note the long resonant decay in the left waveform of **Fig. 4B**. The sinusoidal decay is a pure exponential; that is, a constant number of dB per time unit (linear dB/ms slope). This waveform takes 4.32ms to decay 10dB, and 25.9ms (130 waveform cycles) to decay 60dB.

This waveform is useful for hearing room acoustics properties; it's resonant enough to produce a distinct tonality (the spectral BW is 0.0245 octave, or 0.391 semitone), but the 60dB decay time of 25.9ms is much shorter than the reverberation time of any room except an anechoic chamber.

The right photo in **Fig. 4B** is the same waveform expanded to a 0.1ms/div time scale. Note the clean attack transient starting at the sinusoidal zero-voltage point. (The slight starting curvature is from the input pink pulse's finite rise time of about 13μs.)



TRANSIENT SOUND VARIATIONS

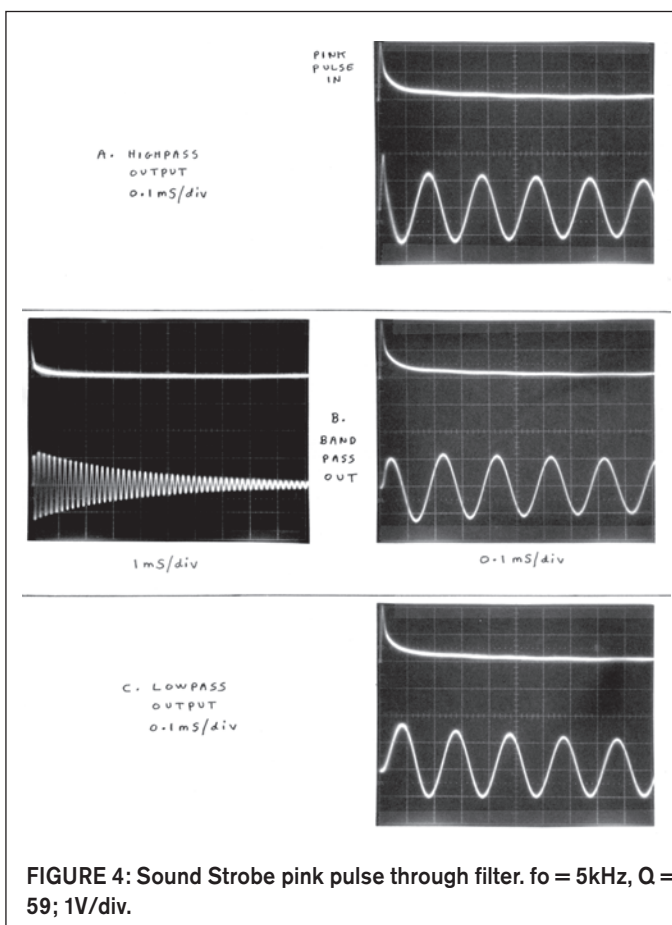
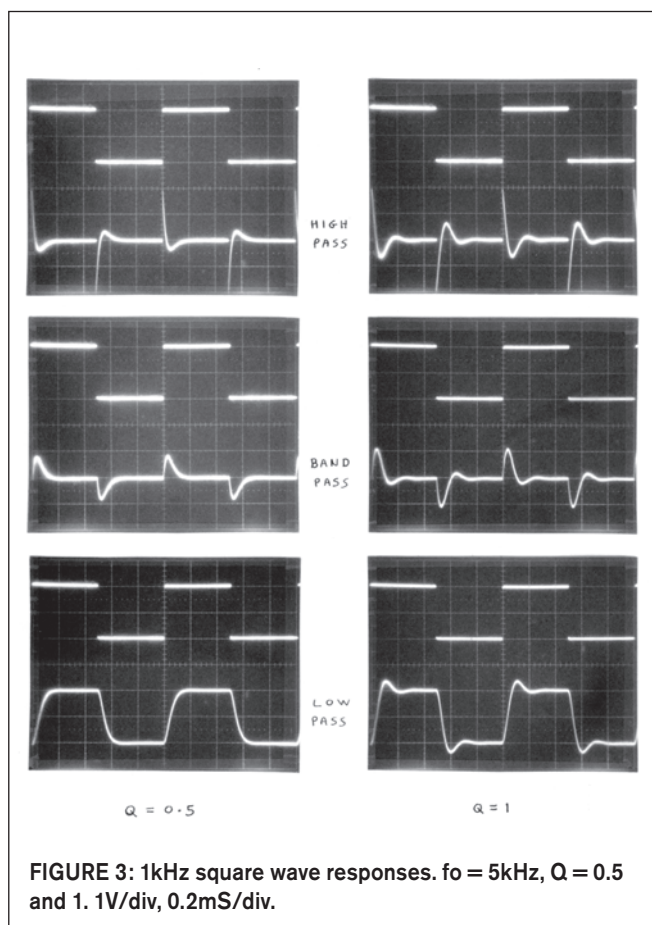
The resonant tone decay is the same at all three filter outputs (except for the $+90^\circ$ HP and -90° LP phase shifts regarding the BP out). But the transient “impact” sounds are quite different. The HP output (**Fig. 4A**) passes the full HF content of the pulse, re-

sulting in a much sharper transient sound (at the 5kHz f_0 shown, the sound starts with a metallic “ting”).

Conversely, the LP output transient is relatively “muted,” but with significant LF content (manifested in **Fig. 4C** by the first half-cycle of higher amplitude than the others).

Figure 5 shows the most extreme range of HF and LF transient spectrum emphasis available. In **Fig. 5A**, the Sound Strobe impulse waveform is used with the filter's HP output. The impulse spectrum (flat per-Hz but upsloping $+3\text{dB/octave}$ on the ear's per-octave analysis) has its strong HF content passed unattenuated by the HP output.

Figure 5B shows the opposite available extreme: The Sound Strobe's 252μs exponential pulse is used. This is a step pulse rolled off at 6dB/octave below 632Hz. The longer pulses (830μs and 25ms exponential) have more extended LF spectra, but I used the 252μs pulse here so its decay could be seen on the fast (0.1ms/div) time scale. The LP filter output is shown; note the strong LF content as the resonant sine cycles have

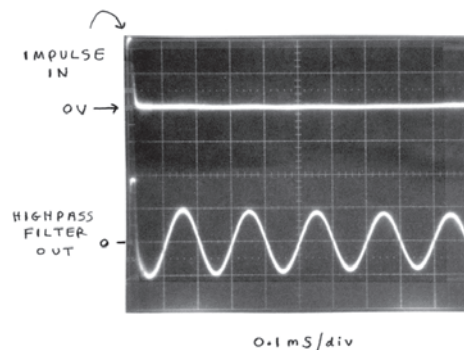


an initial offset that follows the input pulse decay. This type of waveform is useful for hearing a speaker's full-band transient coherence in the presence of a tunable, musical-sounding HF resonant decaying tone.

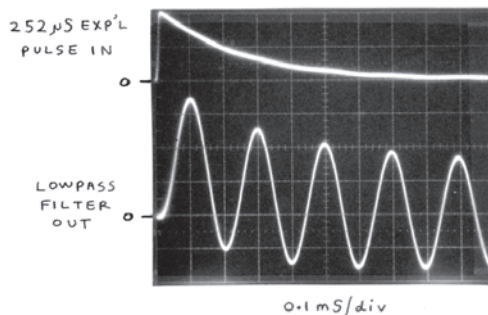
GENERAL PROPERTIES OF FILTERED TRANSIENT PULSES

Figure 6 shows a pink pulse through the BP output, with $f_0 = 1\text{kHz}$ and a Q of 4.318, which corresponds (with a BP filter having $\pm 6\text{dB/octave}$ asymptotic slopes) to a $1/3$ octave BW. Also shown are equations for decay times, for various decay levels, f_0 , and Q values.

Only a handful of pulse outputs have been shown. But consider that with the Sound Strobe and this filter, the range of combinations includes six input pulse shapes, three filter outputs, a filter f_0 range of $20\text{Hz} - 20\text{kHz}$, and a Q range of 0.5 to 30 (manual), 7.3 to over 100 (auto tracking mode). Even assuming that significant differences are represented by $1/3$ octave f_0 steps and 2:1 Q changes, the number of combinations would be 1584. But I forgot to include



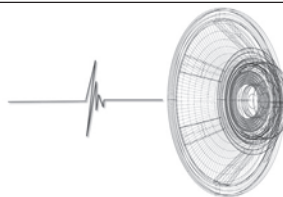
A.
STRONGEST
HIGH FREQ.
TRANSIENT



B.
STRONGEST
LOW FREQ.
TRANSIENT

FIGURE 5: Sound Strobe pulse/filter combinations for strong HF and LF transients at start of resonant tone decay. Filter $f_0 = 5\text{kHz}$, $Q = 59$; 1V/div .

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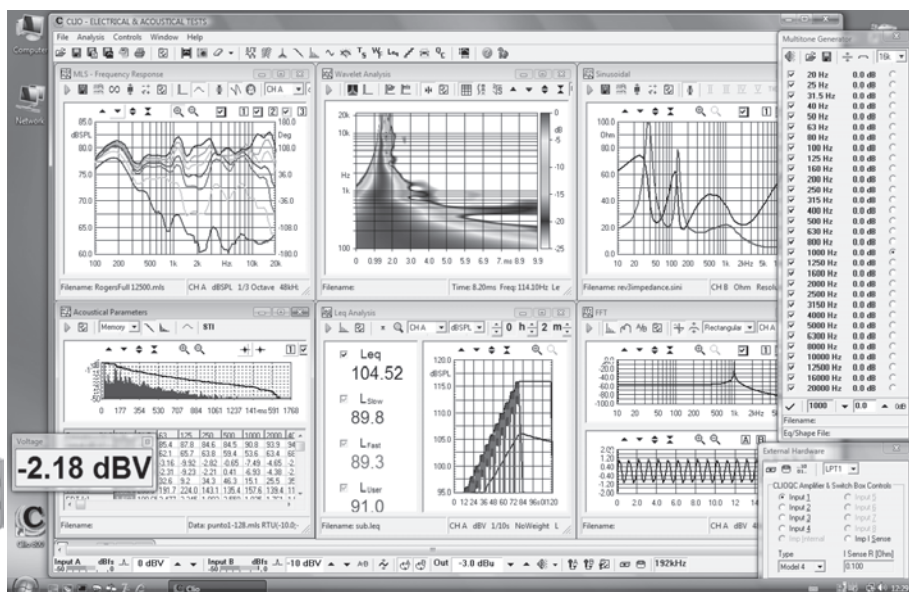


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THE +1.8DB/OCTAVE SLOPE EQ

For the "Auto" Q mode, I experimentally determined the best-sounding amount of Q tracking with f_0 . Roughly a compromise between constant Q and constant decay time (which would be a Q proportional to $\sqrt{f_0}$), the relation turned out to be approximately $Q = 7.3(f_0/20\text{Hz})^{0.4}$. The pulses have enough resonant sustain to resemble musical tone transients, but with short enough decay times to reveal speaker anomalies in transient precision. Also, room effects are heard, distinguishable from the direct speaker output.

The Auto Q mode is meant to be used with the Sound Strobe 25ms exponential pulse, of frequency below about 4Hz (or a square wave below about 2Hz, with 2.3Vpp maximum amplitude, that of the Sound Strobe). These wideband step pulses produce a constant peak

pulse voltage versus f_0 from the filter's BP output. (This is also true at the other outputs and regardless of Q , manual, or auto.)

Due to the decay time shortening as f_0 increases, the energy per pulse (and therefore the RMS voltage for a given pulse repetition frequency, or PRF) decreases with increasing f_0 . The pulse energy decreases with a slope of about -1.8dB/octave (18dB decrease from 20Hz to 20kHz f_0).

The slope EQ, when switched in with S1, compensates for this when a constant pulse energy is desired, such as for measuring the reverberant power/frequency distribution in a room. But there's a disadvantage: As Table 1 shows, the peak BP output pulse voltage increases by 19dB from 20Hz to 20kHz f_0 (0.50V to 4.59V with a 2.0Vpp Sound Strobe 25ms exponential pulse at rates below 2Hz). This can clip amplifiers and possibly blow tweeters.

However, for general speaker/room

evaluations, you should use the constant-peak output with the EQ off (S1 set to "FLAT"). The pulse loudness versus f_0 sounds reasonably balanced.

FILTER CIRCUIT

The circuit (Fig. 8) follows the topology of Fig. 1. U1 is the input and feedback (FB) summing amp, and U2A, U2B are

the integrators. VCA1 and VCA2 are the "a" elements (controlling f_0), while VCA3 is the "b" element (controlling Q). Note (from Fig. 1) that $Q = 1/b$, so if the gain of b is zero, the Q is infinite; that is, you have an oscillator. This is because without b , there are two integrators cascaded in a FB loop. At DC the FB is inverting, but for any AC signal the integrators have a phase shift of -90° (-180° for the two cascaded). So at the frequency (f_0) where the loop gain (determined by a^2 and the value of C6, C12) is unity, the net positive FB produces oscillation.

This is the principle I used in "A Wide-Range Audio Sweep Oscillator" (aX Feb. '07). The operational transconductance amps I used there (the RCA/Harris CA3280AE) are no longer available. Compared to those, the Analog Devices SSM2018TPZ units I used in the present filter for voltage controlled amplifier/attenuator (VCA) devices have some disadvantages and advantages.

Disadvantages:

1. Less BW, about 250kHz instead of 3MHz for the CA3280AE (the latter at maximum gain where high BW is needed because f_0 is proportional to VCA gain). The SSM2018's 250kHz rolloff produces a phase lag (about -5°) at 20kHz. This would increase the integrators' desired -90° phase, which would make the filter oscillate at high intended Q values. C5 and C11 provide a phase lead, compensating this effect.
2. The SSM2018 has a low-Z voltage source output, while the CA3280 has (or rather, had) a high-Z current

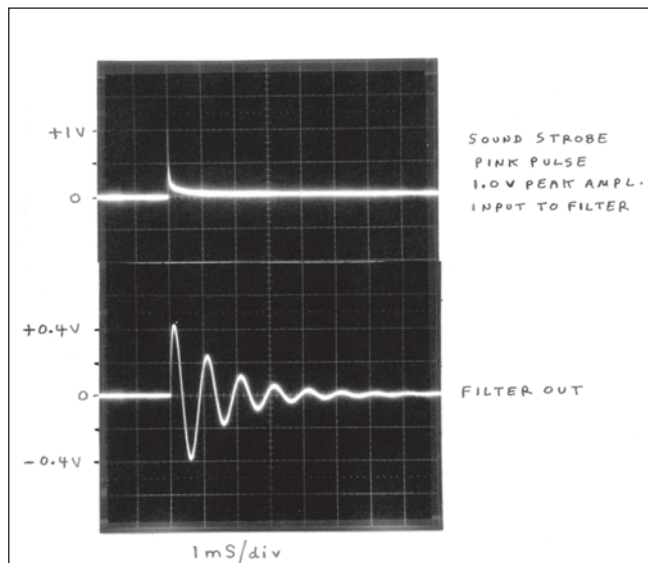


FIGURE 6: Sound Strobe pink pulse through 1/3 octave 1st order bandpass filter, $f_0 = 1.00\text{kHz}$, $Q = 4.318$, $AV(f_0) = 4.318$, BW -3dB = 890.9 - 1122.5Hz.

Time	Decay to:
0ms	0dB
1.375ms*	-8.69dB
1.583ms	-10dB
3.165ms	-20dB
9.495ms	-60dB

* Exponential decay time constant

For any filter f_0 (1kHz shown above): constant pulse energy, $T(-10) = 10\text{dB}$ decay time = $1583/f_0 \text{ ms} = Q/2\pi f_0 \log_{10} e = 0.36647Q/f_0$

$V_p = 0.42V \sqrt{f_0/1000}$, BW -3dB = f_0/Q

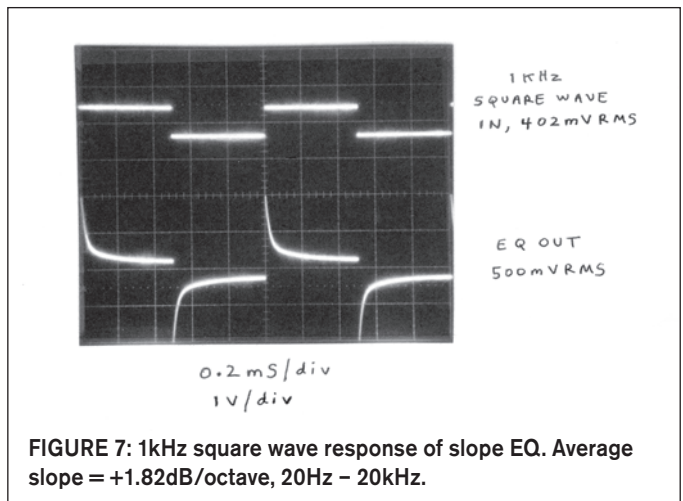


FIGURE 7: 1kHz square wave response of slope EQ. Average slope = +1.82dB/octave, 20Hz - 20kHz.

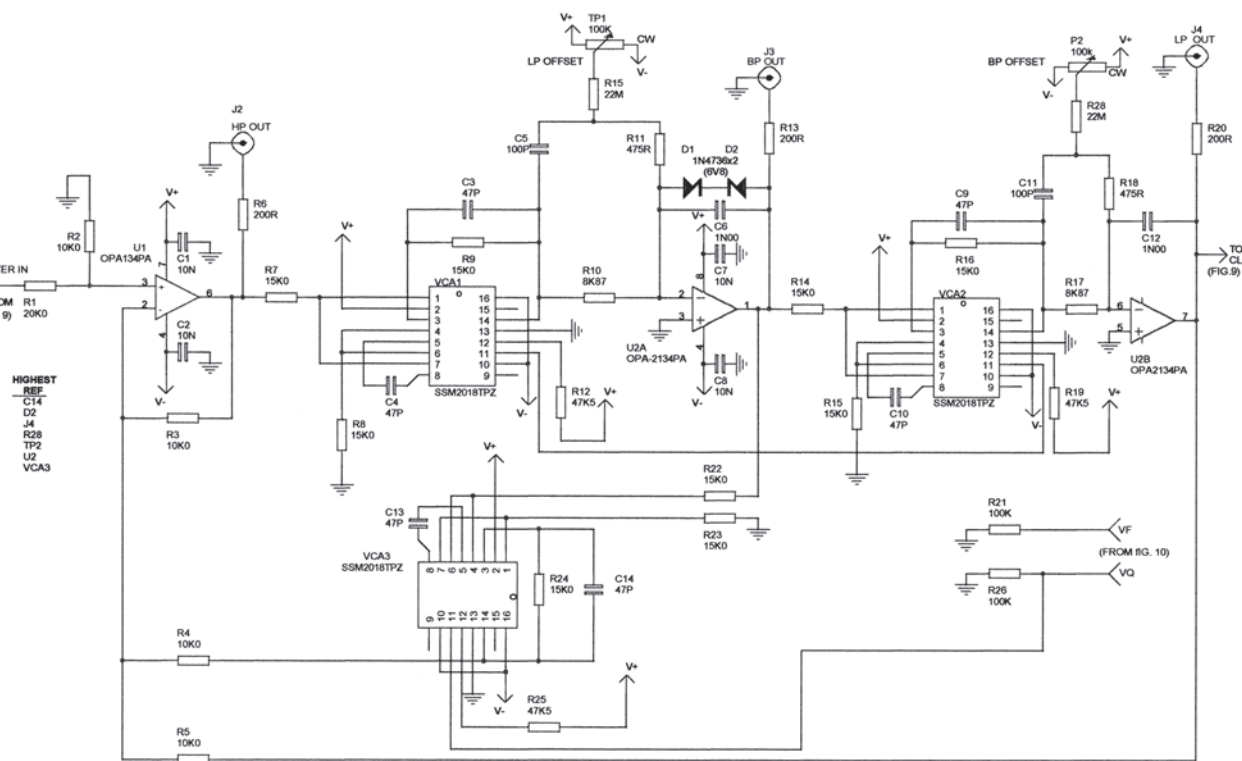


FIGURE 8: Multimode active filter.



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source output. With the latter, DC offsets decrease in proportion to the controlled signal current; also, the integrator op amps' offset voltages don't see a low Z from the VCA. But with the SSM2018's low-Z output, plus the device's own output offset (1mV typical, 15mV maximum), there's a much more potential offset problem.

Because the integrators are open-loop at DC, it's only the filter's overall negative FB that keeps offsets in check. But at low fo values, the VCAs (1 and 2) are at high attenuation (fo is 17.7kHz when the VCA is at unity gain). At 20Hz, the two VCAs combined have 118dB of attenuation (a voltage loss of 787,000:1). So there's much less overall negative FB to control the integrators' output DC offset. TP1 and TP2 are needed to trim the filter for acceptably low offset at low fo (to be described later).

Advantages:

1. The SSM2018 has much lower dis-

tortion (0.01%) and higher gain-control precision, and a 140dB gain-control range.

2. It doesn't need (as the CA3280 did) an external exponential converter; the VCA has a very linear dB versus control voltage relation (at pin 11).

Q CONTROL

VCA3 (the "b" element in Fig. 1) introduces a second negative FB loop, this one around only the first integrator (U2A). This FB produces damping in proportion to the gain of VCA3. That's why the Q is inversely proportional to VCA3's gain. The gain of VCA3 varies from 2 (+6.0dB) for a Q of 0.5, to about 0.0085 (-41.4dB) for the maximum Q of 118 (at 20kHz in "Auto" Q mode).

The zeners (D1, D2) prevent high signal voltages from overdriving the VCA.

SLOPE EQ AND CLIP LIGHT

Figure 9 shows the simple, switchable RC circuit that provides the +1.82dB

TABLE 1: BP output pulse characteristics, "Auto" Q mode. Source is Sound Strobe 25ms exponential pulse, 1Hz rate, 2.0Vpp amplitude. T-20 is 20dB decay time, N-20 is number of waveform cycles to 20dB decay.

fo	Q	EQ gain dB	EQ on		EQ off		T-20 ms	N-20 cycles
			pulse energy dB re 1kHz	peak volts	pulse energy dB re 1kHz	peak volts		
20Hz	7.29	-10.54	-0.25	0.50	+9.64	1.66	267	5.3
50	11.44	-8.60	-0.33	0.61	+7.62	1.69	168	8.4
100	15.83	-6.47	+0.20	0.75	+6.02	1.74	116	11.6
200	21.51	-4.93	+0.06	0.96	+4.34	1.77	78.8	15.8
500	31.07	-2.98	-0.37	1.25	+1.96	1.77	45.5	22.8
1kHz	39.61	-0.65	0.00	1.66	0.00	1.81	29.0	29.0
2	48.5	+1.80	+0.32	2.30	-2.13	1.81	17.8	35.5
5	59.1	+4.85	+0.25	3.34	-5.25	1.81	8.66	43.3
10	71.7	+6.73	-0.04	4.13	-7.42	1.81	5.26	52.6
15	86.0	+7.36	-0.38	4.38	-8.39	1.81	4.20	63.0
20kHz	117.7	+7.62	-0.01	4.59	-8.28	1.81	4.31	86.3

EQ gain is 0dB at 1.191kHz

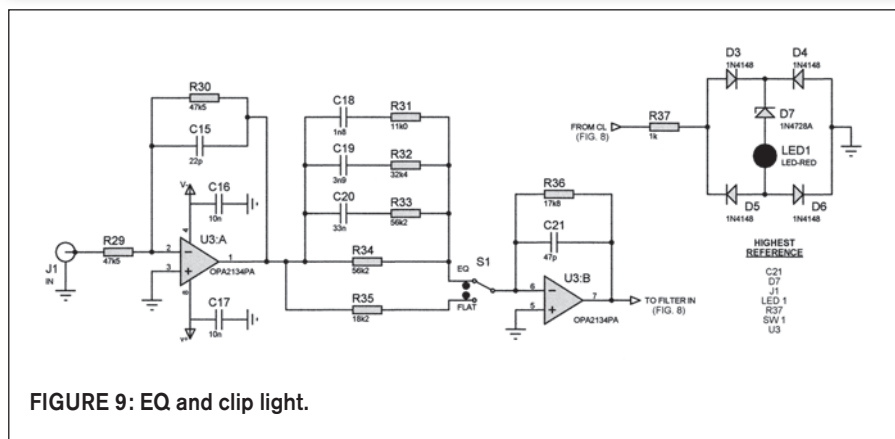


FIGURE 9: EQ and clip light.

octave upslope, reasonably uniform over the audio band. The clipping indicator light (LED1) is useful when the filter Q is high. High Q is mainly intended for very low frequency (0.5-2Hz) Sound Strobe pulses, where you want to hear individual resonant-tone pulse responses of rooms or speakers.

However, the Sound Strobe pulse frequency goes up to 41.2Hz (the low open-E string on a 4-string bass). The filter, with high Q, can tune individual harmonics out of Sound Strobe pulses above about 5Hz, but the filter's gain is numerically equal to Q. Therefore, the filter outputs can easily clip in such applications. LED1 allows you to decrease the Sound Strobe's (or other source's) signal amplitude, when necessary to avoid filter clipping.

CONTROL CIRCUIT

First, notice that this circuit (Fig. 10) uses voltages labeled "+5V PTAT" and "-5V PTAT." "PTAT" stands for "Proportional To Absolute Temperature" (degrees Kelvin). This will be described later; for now, suffice it to say that this temperature compensates for the VCAs' gain-control slope. For constant gain (or attenuation), the VCA control voltage applied must be PTAT.

Switch S2 selects fo (when S3 is in the "oct ±1 oct" position) in octave steps, from 40Hz to 10kHz. Control pot P1 adds a ±1 octave sweep range, so S2 and P1 can control fo from 20Hz to 20kHz.

When S3 is in the "Full Δfo sweep" position, S2 has no effect, but P1 now tunes fo over the full 20Hz – 20kHz range.

PTAT CIRCUIT

The VCAs have a control port sensitivity specification of -30mV/dB at +25° C; I measured -31.7mV/dB. But this is proportional to absolute temperature (PTAT). Therefore, to obtain temperature independence of the VCAs' gains (and thus the filter's fo and Q), the control voltages must be PTAT.

You could accomplish this by using a +3500 ppm/° C resistor (close enough to the ideal 3354 ppm/° C tempco), such as the model PT146 1k unit from Precision Resistor Co. of Largo, Fla. that I used in my 6-channel volume/balance control (aX Feb. '03), which uses the

same VCAs. But these cost \$5 each, the filter would need two (fo and Q), and I know of only this one source.

But there's another way: Make the fo and Q control and bias DC voltages PTAT. With the use of external fo and/or Q control signals, the above TC resistor would be needed. Here, though, the internal control pots and switches can all be supplied from PTAT DC references.

An excellent source for this is the plain old ubiquitous 1N4148 diode. If biased from a constant-current source, and the diode drop is subtracted from an appropriate constant voltage, the resulting voltage can be very close to PTAT, at least over a 0° to +50° C range (+32° to +122° F). The subtraction makes the tempco positive.

I measured the 0° to +50° C tempco of the voltage across three 1N4148 samples, at 1mA. The tempcos were within 1.1% of -2.1433mV/° C. The diode drop average was 611.38mV; highest and lowest were 616.8mV and 605.8mV.

To obtain a PTAT voltage, the output should be the product of the tempco magnitude and 298.15° K (equal to +25°

C), which is 639.02mV. Then the DC reference should be the sum of this plus the typical 1N4148 drop of 611.38mV (at 1mA); this is 1.2504V ideally (the exact value isn't that critical).

By a strange Colincidence (term invented by my father, Robert Colin), the internal reference voltage in an LM317 regulator is very close to 1.25V. (Does National Semi know something I don't?

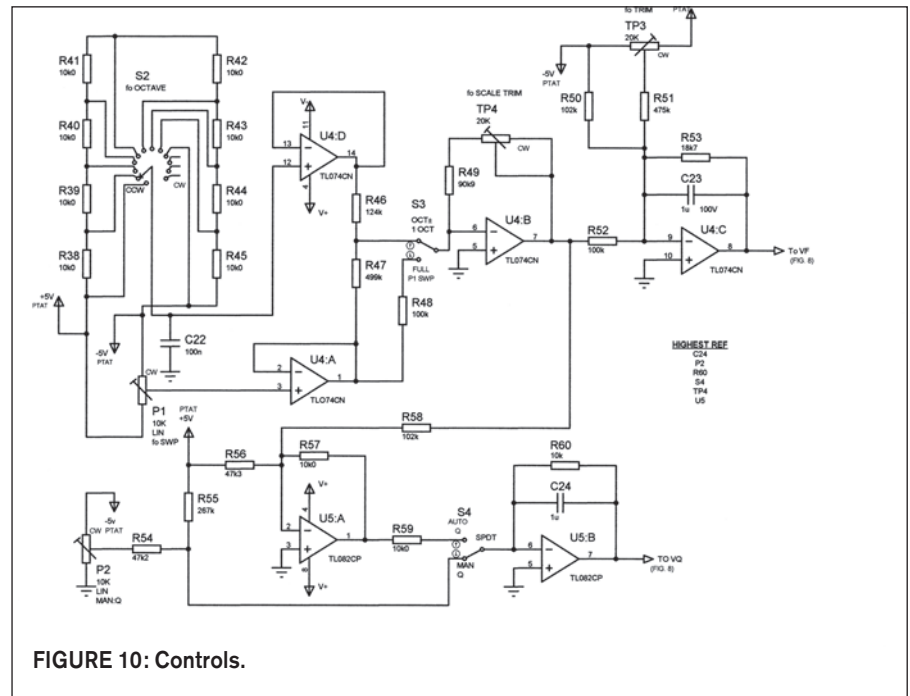
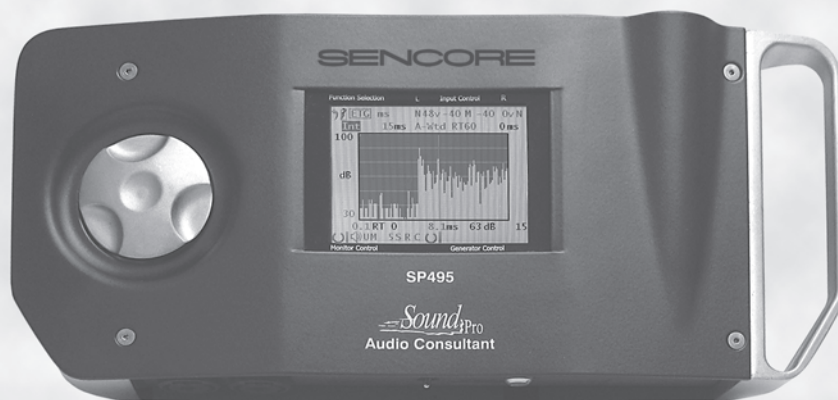


FIGURE 10: Controls.

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The circuit of **Fig. 11** takes advantage of this. The output of the LM317 is always 1.25V (typical VREF) above the ADJ terminal. So the diode, biased negatively regarding ground, has its voltage drop subtracted from the 1.25V VREF; and without the usual external feedback needed, the LM317 output provides a very stable voltage of about +639mV at +25° C that's proportional to absolute (° K) temperature. The 100Ω load resistor ensures that the LM317's minimum load current (3.5mA typical, 5mA maximum) is drawn.

POWER SUPPLY

The circuit in **Fig. 11** is similar to that in the Sound Strobe. The rechargeable 9V batteries can be either NiCd (such as Mouser 639-N6PT) with 110mAH capacity, or the newer NiMH types (such as Mouser 573-GP17R8H) with 170mAH capacity. The latter type is recommended, both for its higher capacity (which will power the filter for about 2.4 hours) and its higher minimum voltage (8.4V as opposed to 7.2V for the NiCd and some other "9V" NiMH batteries).

If you prefer, you can power the filter from any ground-referenced supply of ±9V to ±15V.

TRIMMING

A. Power Supply Test:

1. With the 24V wall supply connected and S5 set to "AC," LED2 ("AC") and LED3 ("PWR OK") should light. Measure the V⁺ and V⁻ voltages regarding ground; they should be in the range of ±9V to ±11V.
2. Measure the +5V PTAT voltages. They should be in the range of ±4.3V to ±5.7V, depending on the forward drop of D13 and ambient temperature.

B. Offset Trimming:

1. Set TP1, 2, 3, 4 to their approximate midpoints.
2. Set S2 to the 630Hz position (the switch's midpoint), P1 to its approximate midpoint, and S3 to the "OCT ± OCT" position. Set P2 (the manual Q pot) fully CW (clockwise), and S4 to "manual Q."
3. Measure the DC voltages at the BP and LP outputs (re ground). If two meters are available, connect one to each output. (Accuracy isn't needed; the cheapest meters that can resolve down to 50mV DC will do.)
4. Lower the "fo octave" setting (S2) one step at a time. At some point the offsets will increase signifi-

cantly. If using only one meter, alternately observe the BP and LP offsets.

5. Lower the "fo octave" setting as many steps as possible without either offset exceeding ±2V.
6. Adjust TP1 to lower the LP offset as much as possible (it might drift around, but you should be able to null it within ±50mV).
7. Adjust TP2 to null the BP offset.

Note: It might seem strange that each trimpot affects not the op amp integrator that it feeds, but rather the opposite one. This is because the integrators' offsets are stabilized not by local op amp FB (there isn't any at DC), but by the overall loop. Each integrator receives DC feedback from the other one's offset.

8. Lower the fo setting further, and repeat steps 6 and 7.
9. Continue this until the fo setting is 20Hz (S2 fully CCW (counter-clockwise) and P1 fully CCW). Then null the offsets as much as possible.

C. Frequency Trimming:

Note: This requires a 20Hz – 20kHz sine wave generator and an oscilloscope (preferably) or an AC voltmeter capable of responding to the 20Hz – 20kHz range (voltage accuracy and true RMS response are not necessary).

1. Connect the audio generator to the filter input. If a dual-channel scope

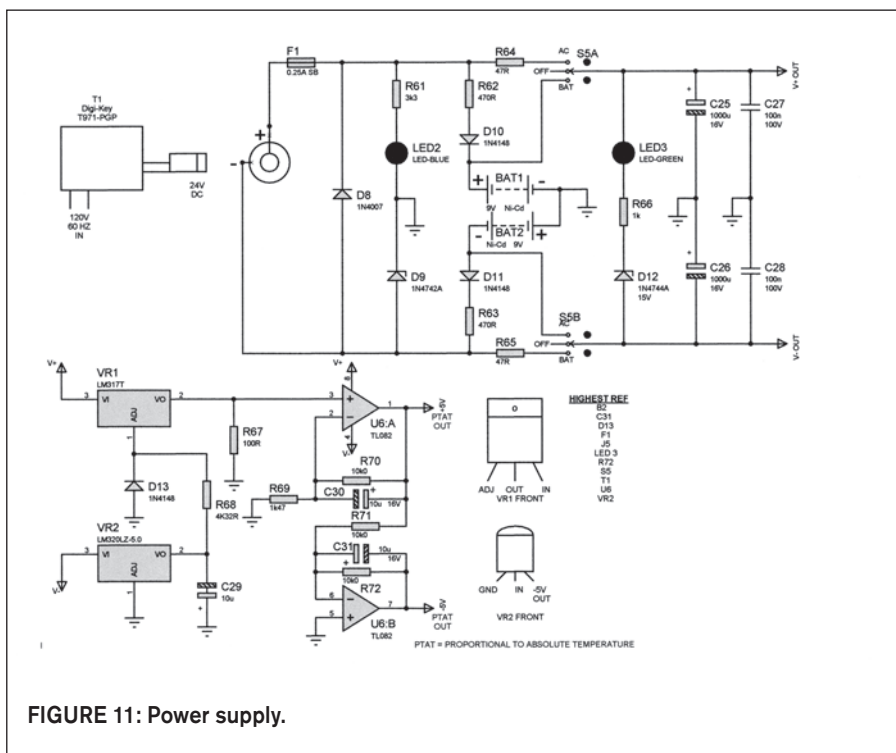


TABLE 2: Filter data (with slope EQ switched to "Flat" position).

Gain and Frequency Response

LP out: DC coupled, gain = 0dB for f << fo, gain = Q at fo, rolloff = -12dB/octave for f >> fo.
BP out: gain = Q at fo, rolloffs = ±6dB/octave for f far from fo.
HP out: gain = Q at fo, rolloff = +12dB/octave for f << fo, HF response -3dB at 3MHz.

Maximum output = ±6.8V peak, 4.8V RMS

Zin = 47kΩ, Zout = 200Ω

THD ≈ 0.01%

fo range = 20Hz - 20kHz

Q range = 0.5 - 30 (manual), 7.3 - 118 (Auto)

Noise Out, RMS

fo = 1kHz, Q = 1: 99μV (20Hz - 20kHz), 62μV ("A" weighted)
fo = 20Hz, Q = 1: 875μV (20Hz - 20kHz), 78μV ("A" weighted)
fo = 1kHz, Q = 30: 250μV (20Hz - 20kHz), 230μV ("A" weighted)

Output DC offset: Trimable to ±100mV

Power Supply

24V DC, approximately 70mA, ground isolated.
May also use ground-reference supply of ±9V to ±15V, approximately ±70mA.

is available, connect the generator also to channel 1. Trigger the scope either from channel 1 internally, or (preferable) externally from the generator's trigger output, if it has one. Connect the scope's channel 2 to the filter's BP output. Set the generator's output amplitude to about 0.5V RMS ($\pm 0.7V_{pp}$ on the scope), and frequency to 630Hz.

2. Not required, but desirable, is to monitor the source frequency with a counter, unless the generator's frequency calibration is accurate to about $\pm 2\%$.
3. Switch the slope EQ to "Flat," S2 to its midpoint (630Hz), S3 to "Oct ± 1 oct," and S4 to "manual Q." Set P1 to its midpoint, and P2 (manual Q) also to its midpoint (Q about 5).
4. Adjust TP3 (fo trim) for a maximum output from the BP out, on channel 2 of the scope (or AC meter if a scope isn't available). The maximum BP output voltage should be about 5 $\times$ the input voltage. Note: the most accurate and sensitive way of tuning is for zero phase shift between the input and output waveforms. And the best way to see this is to switch the scope mode to "X-Y" if available; then at zero relative phase the ellipse will collapse to a straight tilted line.
5. Change the generator frequency to 20Hz, and set the filter fo controls to 20Hz by rotating both S2 and P1 fully CCW.
6. Adjust TP4 (fo scale trim) for maximum filter output and, if possible, zero input/output phase (as in step 4).
7. Set the generator frequency to 20kHz, and the fo controls also to 20kHz (S2 and P1 fully CCW).
8. Adjust TP4 for tuning as in steps 4 and 6.
9. For the most accurate full-band tuning, alternately repeat steps 6 and 7, while periodically adjusting TP3 if necessary to offset both the 20Hz and the 20kHz settings. This can be somewhat laborious, but is needed only if the highest accuracy is desired. For most applications, the results of step 8 are acceptable.

HIGHEST Q STABILITY

The loop phase-compensating caps C5 and C11 have a strong effect on stability at 20kHz and maximum Q (which occurs in the "Auto" Q mode). To test this stability:

1. Connect a 50Hz (approximately) 1Vpp square wave to the filter, and monitor the LP output on a scope.
2. Set fo to 2.5kHz. Set the Q mode to "manual," and set P2 (manual Q) fully CCW (Q = 0.5). The output should be a LP filtered square wave, with no overshoot.
3. Turn up the Q control slowly to fully CCW. As you do, decaying ringing cycles should appear.
4. Switch S4 to "Auto" Q mode. At fo settings above 1kHz (such as the 2.5kHz it's set to now), the Q should increase when switching S4 from "manual" to "Auto." This should make

the ringing's decay longer.

5. Set P1 to about two-thirds of the way up, and switch S3 to "Full P1 Sweep."
6. Slowly turn up P1. When it's up fully CW, the filter might break into continuous oscillation at 20kHz. If this happens, C5 needs to be increased. Increasing it from 100pF to 120pF should be enough. Increasing it too much will reduce the maximum available Q.
7. The ideal is to have a value of C5 (C11 has the same effect) that produces a long ringing decay at 20kHz fo, without oscillating. C5 can be a trimmer cap with a minimum value of 80pF or less, and a maximum value of at least 150pF.

Table 3 is the parts list. **Figure 12** shows my suggested enclosure panel component layout. Please let *audioXpress* know

SPEAKER CROSSOVER COMPARISON EXAMPLE

I found the filtered Sound Strobe pulses to be very revealing of a speaker CO alignment. I could easily hear the frequency delay dispersion (see my article by that title in *aX* 11/07) of a woofer/mid CO that had very flat amplitude/frequency response, the Swans M1/Scan-Speak sub unit ("Stereo Subs for the Swans M1," *aX* Sept. '05).

The raw Sound Strobe pulses, particularly the LF, 25ms exponential, and 830 μ s exponential pulses, clearly revealed the 220Hz CO's transient delay dispersion. But filtering the 25ms exponential pulse with the BP filter fo at 220Hz made the delay dispersion more audible. I compared the Swans system to a full-range Seas coincident driver in a sealed box. With a filter Q of 30, the Swans made an audible separation between the transient attack and the resonant tone. But with the Seas, the sound was coherent; I could picture an actual drum skin sound.

Interestingly, there were many positions in the room where I could hear a standing wave null, almost removing the pulse's 220Hz tonality, with both speakers. However, I could still hear the superior "live drum-like" transient response of the full-range Seas, *regardless of room effects*.

This correlates with recorded drums and string bass sounds, but I must have to listen for the Swans/sub delay-smearing effect. With the filtered Sound Strobe pulses, it's immediately recognizable.

Figure 13 shows an electronic representation of this effect: The "A" waveform is the 25ms exponential pulse, filtered with fo = 232Hz and Q = 0.65; that Q produced the cleanest pulse without significant overshoot.

Waveform "B" is this pulse through a three-stage nonresonant allpass network, with its fo (center of its phase/frequency graph on a log frequency scale) at 232Hz, the same as the BP filter's fo. The allpass network's group delay ranges from 4.12ms for frequencies much lower than 232Hz, to 2.06ms at 232Hz, and approaches zero at frequencies much above 232Hz.

This allpass response is similar to a fourth-order CO that (like the allpass) has a flat amplitude/frequency response. Note how the clean transient "A" waveform is smeared with frequency selective delay. The original waveform's peak is delayed about 3.4ms, and also reduced in amplitude. *Unlike some measurements, this effect is as audible as it looks.*

whether you would find this filter useful. If there's enough interest, a kit and assembled product may become available. **Table 4** shows bandwidth versus Q data.

CONCLUSION

The wide range of transient pulses shown in this article can be very useful test signals for directly hearing speaker time misadjustment, frequency response anomalies, and spatial image coherence; and also the effects of room acoustics. The principal features are:

1. The raw Sound Strobe pulses are best for hearing wideband transient response and overall freedom from (or degree of) speaker coloration, and the smoothness of room acoustics (freedom from discrete echoes).
2. The filtered pulses allow frequency tuning of all the audible effects described. The Q control allows a wide range of resonant tonality (including none) to be selected.
3. The precise-transient resonant tone pulses, because of their resemblance to natural acoustic sounds, provide

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TABLE 3: Parts List

Ref.	Description	Size	Mfr	Mfr/P/N	Dist.	Dist. P/N
B1, 2	battery, 9V, NiMH				M	573-GP17R8H
C15	cap, 22pF cer.					
C3, 4, 9, 10, 13, 14, 21	cap, 47pF cer.					
C5	cap, 100pF cer.					
C11	cap, 120pF cer.					
C6, 12	cap, film, 1nF, 100V				DK	PS1102J
C18	cap, 1n8, 100V				DK	PS1182J
C19	cap, 3n9, 100V				DK	PS1392J
C1, 2, 7, 8, 16, 17	cap, 10nF, 100V				DK	PS1103J
C20	cap, 33nF, 100V				DK	PS1333J
C22, 27, 28	cap, 100nF, 100V				DK	EF1104
C23, 24	cap, 1μF, 100V				DK	EF1105
C29-31	cap, 10μF, 16V				DK	493-1279
C25, 26	cap, 1000μF, 16V				DK	493-1287
D1, 2	zener, 6.8V	1W		1N4736A-T	DK	1N4736ADICT
D3-6, 10, 11, 13	diode			1N4148		
D7	zener, 3.3V	1W		1N4728A-T	DK	1N4728ADICT
D8	diode			1N4007-T	DK	1N4007ADICT
D9	zener, 12V	1W		1N4742A-T	DK	1N4742ADICT
D12	zener, 15V	1W		1N4744A-T	DK	1N4744ADICT
F1	fuse, 0.25A, SB					
J1-4	jack, RCA				PE	091-1120
J5	jack, DC in, isolated				M	163-4303-EX
LED1	LED, red				DK	67-1612
LED2	LED, blue				DK	P466
LED3	LED, green				DK	516-1358
P1, 2	pot, 10k lin				DK	RV4N103C
R64, 65	Res, 5%, 47R	¼W				
R67	Res, 100R	¼W				
R62, 63	Res, 470R	¼W				
R37, 66	Res, 1k0	¼W				
R61	Res, 3k3	¼W				
R27, 28	Res, 22M	¼W				
R6, 13, 20	Res, 1%MF, 200R	¼W				
R11, 18	Res, 475R	¼W				
R69	Res, 1k47	¼W				
R68	Res, 4k32	¼W				
R10, 17	Res, 8k87	¼W				
R2-5, R38-45, R57, 59, 60, 70-72	Res, 1%, 10k0	¼W				
R31	Res, 11k0	¼W				
R7-9, 14-16, 22-24	Res, 15k0	¼W				
R35, 36	Res, 17k8	¼W				
R53	Res, 18k7	¼W				
R1	Res, 20k0	¼W				
R32	Res, 32k4	¼W				
R54	Res, 42k2	¼W				
R12, 19, 25, 29, 30, 56	Res, 47k5	¼W				
R33, 34	Res, 56k2	¼W				
R49	Res, 90k9	¼W				
R21, 26, 48, 52	Res, 100k	¼W				
R50, 58	Res, 102k	¼W				
R46	Res, 124k	¼W				
R55	Res, 267k	¼W				
R51	Res, 475k	¼W				
R47	Res, 499k	¼W				
S1, 3, 4	Switch, SPDT				DK	
S5	Switch, DPDT, Ctr, Off				DK	
S2	Switch, rotary, 12 pos.				M	105-SR2611F-26-21RS
T1	Wall transformer, 24V DC	400mA			DK	T971-P6P
TP3, 4	Trimpot, 20k				DK	32967-203
TP1, 2	Trimpot, 100k				DK	3296Y-104
U1	op amp		TI	OPA134PA		
U2, 3	op amp, dual		TI	OPA2134PA		
U4	op amp, Quad		TI	TL074CN		
U5, 6	op amp, dual		TI	TL082CP		
VCA1-3	voltage control amp		AD	SSM2018TP2		
VR1	Reg., Adj. pos			LM317T		
VR2	Reg., -5V			LM320LZ-5.0		
	LED Lens, red				DK	L30001
	LED lens, blue				DK	L30006
	LED lens, green				DK	L30005
	Knob, 0.925" diameter				DK	226-4012
	Knob, 0.750" diameter				DK	226-4011
	Battery holder				DK	BH9V-VW
	Fuse holder				DK	F006

DK = Digi-Key, PE = Parts Express, M = Mouser

good correlation with musical reproduction quality, but with very high sensitivity and repeatability. *aX*

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1. Dennis Colin, "The Sound Strobe," *aX* March '06.

2. Ed Simon, "Product Review: The Sound Strobe," *aX* April '07.

3. Dennis Colin, "A Wide-Range Audio Sweep Oscillator," *aX* Feb. '07.

TABLE 4: Normalized and octave bandwidths as a function of Q

Q	Lower* f - 3dB	Upper* f - 3dB	-3dB BW*	-3dB BW octaves
0.5	0.4142	2.4142	2.0000	2.5431
0.7071	0.5176	1.9319	1.4142	1.9000
1	0.6180	1.6180	1.0000	1.3885
1.4142	0.7071	1.4142	0.7071	1.0000
2	0.7808	1.2808	0.5000	0.7140
2.8710	0.8409	1.1892	0.3483	1/2
4.3185	0.8909	1.1225	0.2316	1/3
8.6514	0.9439	1.0595	0.1156	1/6
14.424	0.9659	1.0353	0.0693	1/10
30	0.9835	1.0168	0.0333	0.04809

*Normalized to $f_0 = 1\text{ Hz}$

1. Note that for $Q = 1$, the upper and lower -3dB frequencies are 1.618034 and 0.618034. These are the "Golden Ratios," exactly equal to $1/2(\sqrt{5} \pm 1)$.

2. The normalized -3dB frequencies ($f - 3\text{dB}/f_0$) are:

$$\frac{1}{2} \left[\sqrt{\left(\frac{1}{Q^2} + 4 \right)} \pm \frac{1}{Q} \right]$$

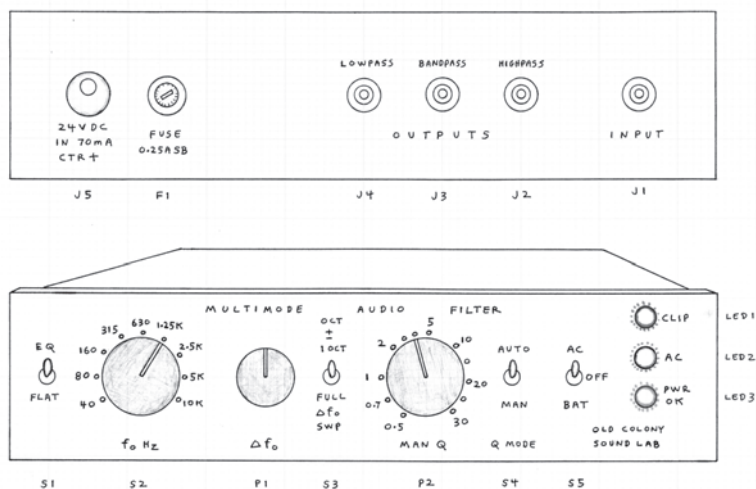
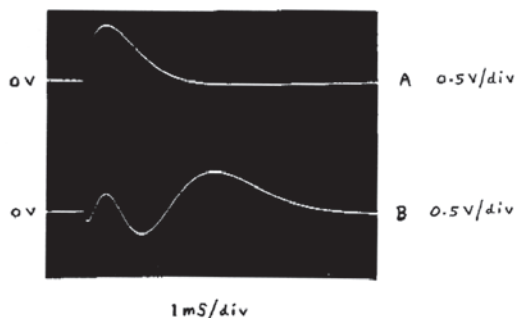


FIGURE 12: Suggested enclosure panel component layout, based on front panel dimensions = 9.0" x 2.1".



A: Sound Strobe 25ms Exp'l Pulse Through Bandpass Filter, $f_0 = 232\text{ Hz}$, $Q = 0.65$

B: Above Through 3-Stage Nonresonant Allpass (Simulating Ideal 4th order Crossover), $f_0 = 232\text{ Hz}$

FIGURE 13: Simulated woofer/mid crossover effect on filtered Sound Strobe pulse.

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