Suppression of Low-Level Impulse Noise*

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A more sensitive system for record impulse-noise suppression has been developed which is effective not only on large pops, but also on very small ticks. Audible damage to the program content is minimized by more rapidly detecting the noise in the ultrasonic region, switching the signal off for microseconds instead of milliseconds, and filling the gap with the low-frequency component.

1. PROBLEMS OF DETECTION AND ELIMINATION

Systems for eliminating ticks and pops during playback of phonograph records designed in the past tend to be effective on large noise impulses only. If the system sensitivity is increased to respond to the small natural ticks and groups of ticks on older records, the damage to the program content exceeds the benefit of the noise reduction. This paper describes a system which is far more sensitive and able to remove many more small ticks, as well as large pops, with very little damage to the program content.

There are two problems basic to the development of a noise-impulse suppressor: detection of the beginning and duration of each individual noise impulse and eliminating the noise from the signal without creating an audible transient. A human being can readily perceive the difference between a tick or pop and the desired program content. An electronic circuit, which can reliably utilize all the clues in the program content long before and long after the tick, becomes unduly complex and expensive. In the system to be described real-time detection occurs based on the very low duty factor of noise impulses as compared with the high-frequency content of the program.

Elimination of individual ticks or pops is not as easy as cutting out a piece of tape and splicing on an angle. Older records may contain a crackling or hashlike sound which is actually a group of ticks occurring as close together as several hundred microseconds. Switching off the entire signal for the duration of the noise burst may produce an unacceptably audible gap.

When a sine wave is turned off and then on again, a plainly audible noise transient is generated. Switching at the optimum phase to reduce the dc content of the transient reduces the audible effect but does not eliminate it. What really needs to be done is to ascertain what the signal would have been in the absence of the noise and fill the gap with the proper signal. If the signal is changing slowly, that is, if it consists solely of low-frequency content, it is not difficult to simulate the actual signal content. However, if the gap is so wide as to encompass many undulations of the signal, it is extremely difficult to predict the true signal content based on its past history or even from the signal values both before and after the tick.

Initial experiments indicated that the shorter the interruption, the less audible it was. Switching the sine wave completely off produced less audible disturbance at high frequencies than a sample-hold circuit which occasionally produced very large disturbances due to holding the wrong values. At low frequencies approaching direct current, the sample-hold circuit was preferable to switching off the signal because the gap was filled with nearly the proper

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signal value. Experiments were also made with fading the signal on and off gradually over a period of milliseconds. Surprisingly the audible effect of the disturbance was greater than that of simply turning off the signal, the reason being that the total disturbance lasted a much longer time. Since most ticks have a duration of less than 200 μs, it was concluded that the least offensive method of switching was to turn off the signal at high frequencies and to sample and hold the signal at low frequencies. Using high-frequency deemphasis after the switch further reduced the audible effects of short gaps in the signal.

The system finally adopted switches the signal for approximately the duration of each tick. Instead of switching the signal off, the gap is filled with the approximate low-frequency content that would have occurred in the absence of noise. High-frequency disturbances are alleviated by high-frequency preemphasis before the switch and complementary deemphasis after the switch. This system is imperfect, but since most ticks produce a gap of only 80–200 μs, large numbers can be removed with negligible audible disturbance. Ticks are detected in real time by means of their high-frequency energy in the 30-kHz region which is compared with the average value of the high-frequency signal energy immediately before the tick. It turned out that magnifying the very-high-frequency energy of record noise impulses showed that these waveforms are not smooth. The roughness in the shape of each noise impulse produces continuous noise output in the 30-kHz region for the entire duration of the impulse. The duration of this measured noise plus a delay of approximately 60 μs is used to determine the off time of the switch. A short delay of 40 μs on the analog signal assures that the leading edge of each tick will be suppressed. It is this short off time which makes it possible to greatly increase the sensitivity of the system to small ticks without destroying the program.

2. SYSTEM DESCRIPTION

A block diagram of the 2-channel stereo system is shown in Fig. 1. In the left signal channel the high frequencies are first preemphasized using a 160-μs time constant or 1-kHz turnover frequency. Overload is prevented by reducing the gain. The signal is next delayed using a 9-pole low-pass filter and phase corrector. At this point the signal branches into two paths. In the lower path the signal passes through a sample-hold circuit and 300-Hz low-pass filter. In the upper path the signal is phase shifted by an amount equal to the phase shift of the 300-Hz low-pass filter, which is approximately 90° at 500 Hz. An electronic single-pole double-throw switch selects the output of the phase shifter for normal signals and switches to the output of the 300-Hz low-pass filter during each tick. The duration of the switchover is 80–600 μs and occasionally up to 2 ms. Following the switch, high-frequency deemphasis restores the signal to flat response typically within ±0.2 dB from 10 Hz to 20 kHz. This very flat response is crucial to the transparency of the system for high-quality signals.

The right channel shown in the lower two rows of blocks is identical with the left channel. To detect the signal, a difference amplifier is connected to the outputs of the two high-frequency preemphasis networks. The signal is then adjusted in level by means of the SENSITIVITY potentiometer and amplified in a 15-kHz to 50-kHz bandpass filter. The signal is then full-wave rectified to direct current using a feedback rectifier circuit for high accuracy. At this point the signal splits into two paths which ultimately feed a comparator. One path feeds a 250-Hz low-pass filter which serves to average the signal and delay the rise when an impulse occurs. The THRESHOLD control adjusts the dc gain of this filter between 6 and 34 dB. A peak rectifier following the low-pass filter stretches the decay of the averaged signal so as to inhibit operation due to overtones of low-frequency musical instruments. Both the output of this peak rectifier and the direct output of the precision full-wave rectifier feed a comparator. After an off delay of approximately 60 μs the square output of the comparator is used to feed the electronic switch in each channel and the sample and hold circuit. In essence, the threshold control adjusts the level of the average signal which the peak signal must overcome to be considered a noise impulse. Musical signals which have much larger duty factors than noise impulses produce sufficient output from the 250-Hz low-pass filter to inhibit the comparator output.

It is apparent from the design of this system that it does not completely discriminate between the onset of a musical signal, such as a castanet, and a noise impulse. When the system does mistake a musical tone burst for noise, it turns off the signal for a very short time, generally a maximum of 1 ms, with no audible effect.

3. PERFORMANCE

The system uses a substantial number of operational amplifiers and precision and selected components to produce extremely flat frequency response, matched channel phase shift, and total harmonic distortion, typically less than 0.05% from 20 Hz to 10 kHz at 2.5 V rms input and output. Noise is 97 dB below rated output.

As noted earlier, the detector system responds to the difference in high-frequency energy between the left and right channels. Therefore the system should be used only with stereo phono cartridges, even when playing monophonic records. The detector responds to the vertical com-

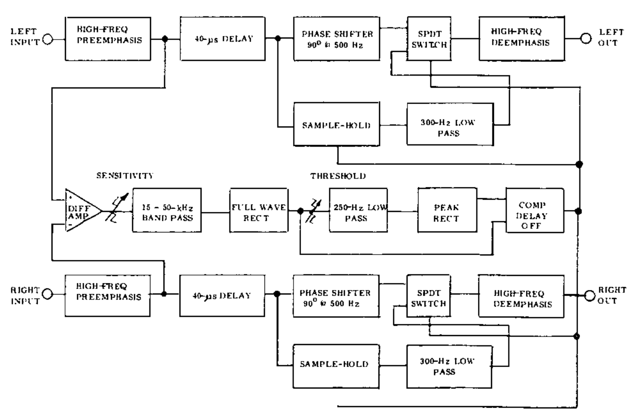


Fig. 1. Block diagram of record impulse-noise suppressor.

ponent of the high-frequency content, and its sensitivity versus frequency is shown in Fig. 2. This curve measured at the output of the 15-kHz to 50-kHz bandpass filter shows a typical gain of 40 dB peaking broadlv at 28 kHz.

Because of the 90° phase shift at 500 Hz and the 40-μs delay in each signal channel, there are considerable changes in the wave shapes of square waves. Fig. 3 shows the effect on a 3-kHz square wave which shows the 40-μs delay from the left side of the picture to the beginning of the leading edge, a small preshoot, and ripple due to the 28-kHz cutoff of the delay network. Fig. 4 shows more clearly the effect of the phase shifter on a 200-Hz square wave. Although the square wave shape is considerably distorted, if there is any audible effect from this phase shifter, it requires considerable imagination to detect it. When the sensitivity and threshold controls are set so that the system responds to the leading edges of the square wave as though a series of noise impulses were present, the switches clip off the spikes as shown in Fig. 5.

When a noise spike occurs in the presence of a low-frequency signal, the system does an excellent job of restoring the signal content during the switch-off time. This effect is illustrated in Fig. 6 which shows a 50-Hz sine wave of 2.8 V peak to peak emerging from the left channel, while at the same time a 2-V peak-to-peak 200-Hz square wave is fed into the right channel, and the sensitivity and threshold controls are set so that the system responds to the leading edges of the square wave as though they were noise impulses. The distortion of the 50-Hz sine wave is barely perceptible. At 2 kHz noise impulses from a 1-Hz square wave fed to the opposite channel have a much greater effect, as shown in Fig. 7. The three different traces commencing at the leading edges of the input square wave show the results of 200-μs switchovers occurring at three different phases of the 2-kHz wave. For these measurements the control settings were typical of those used for playing records. In this instance the very large am-

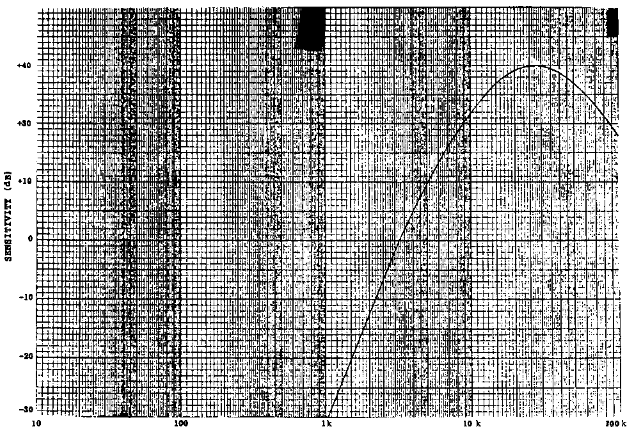


Fig. 2. Detector frequency response. Measured at bandpass filter output with input to one channel.

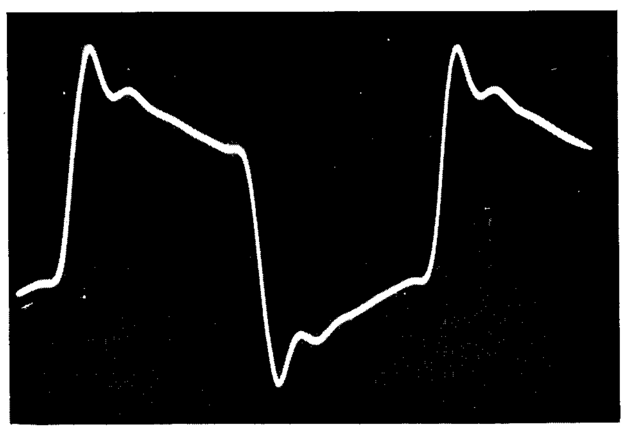


Fig. 3. 3-kHz square wave response showing the effects of the 40-μs delay filter and phase shifter. The trace was triggered by the input.

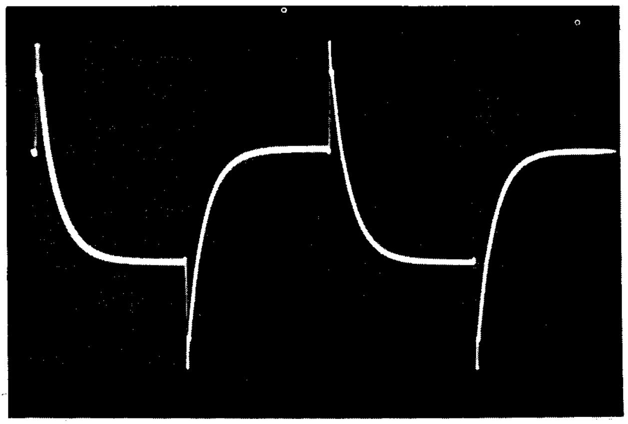


Fig. 4. A 200-Hz square wave is distorted by the phase shifter.

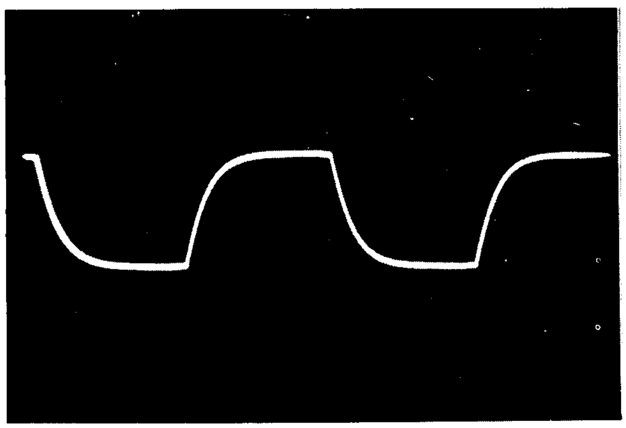


Fig. 5. When the controls are adjusted to detect a 200-Hz square wave, the leading edges are clipped off.

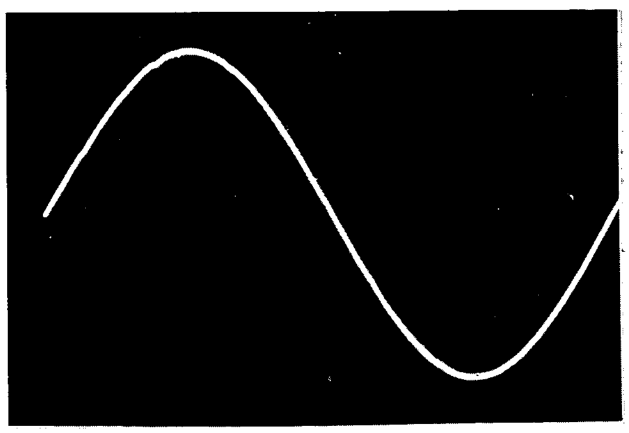
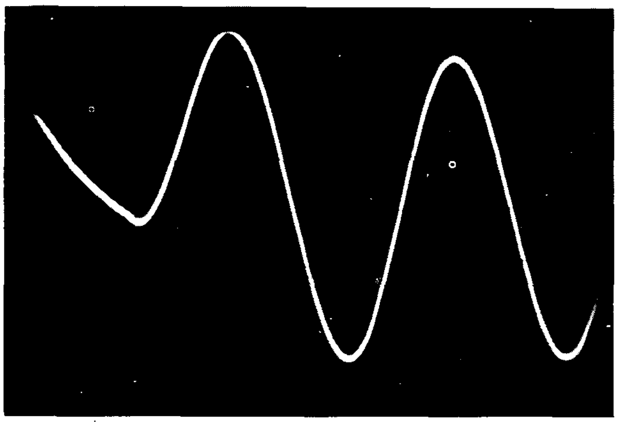


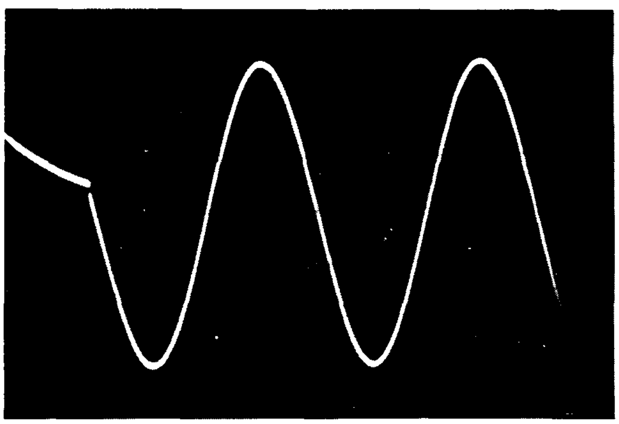
Fig. 6. 2.8-V peak-to-peak 50-Hz sine wave showing the excellent restoration of the signal during switching times. A 2-V peak-to-peak 200-Hz square wave fed to the opposite channel triggered switchovers lasting 200 μs each.

plitude leading edge of the square wave caused the switch-off time to be approximately 200 μs. Smaller noise impulses produce shorter switching times.

When a tone burst at 1 kHz is fed into the system, there is a distortion of the waveforms due simply to the 90° phase shift at 500 Hz and the low-pass filter used for the 40-μs delay. Fig. 8 shows the tone burst when the system is switched out and acts as a unity gain buffer amplifier. Fig. 9 shows this waveform distortion with the system switched in, but the sensitivity and threshold controls are set at zero so that the electronic switch is not activated. At the typical settings of these controls the system detects the leading edge, and there is a slight modification of the first half-cycle of the waveform as shown in Fig. 10. At 5 kHz there is also distortion of the waveform due to the 28-kHz low-pass filtering and phase shift as shown in Fig. 11. When the controls are set at their normal positions for record production, the system does mistake the leading edge of the tone burst for noise and switches off the first 1 ms of the signal shown in Fig. 12. This is an improvement over other available systems which show 2.5–4 ms lost in such a test.



(a)



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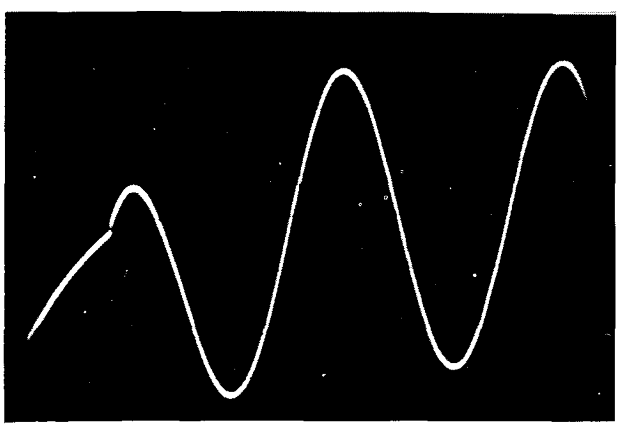

(c)

Fig. 7. At 2 kHz the switching effect is quite visible. The three single traces were triggered by the leading edges of a 1-Hz square wave fed to the opposite channel. A 200-μs period of distortion is shown at three different phases of the 2-kHz wave.

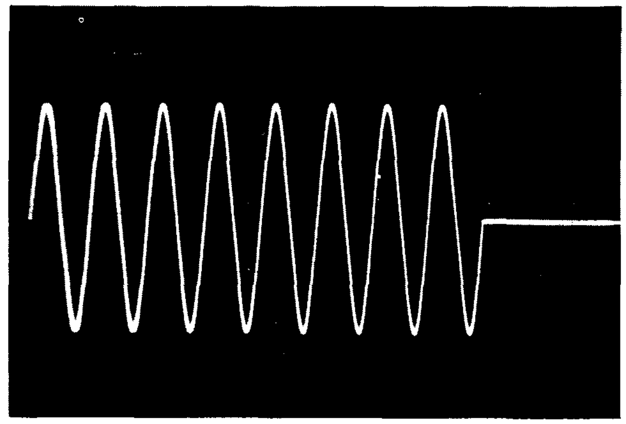

Fig. 8. 1-kHz tone burst response of the unity gain input buffer amplifier.

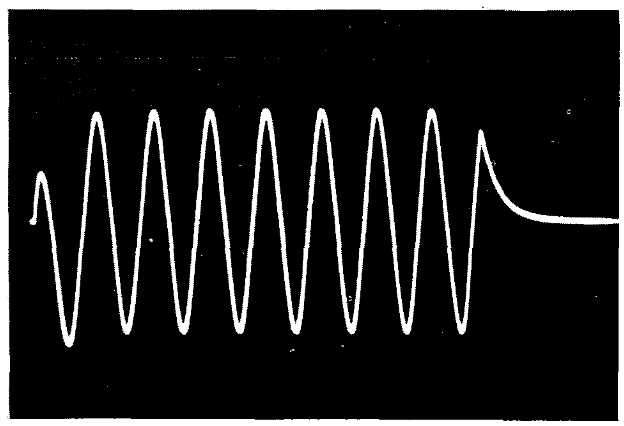

Fig. 9. Effect of the phase shift on a 2-V peak-to-peak 1-kHz tone burst.

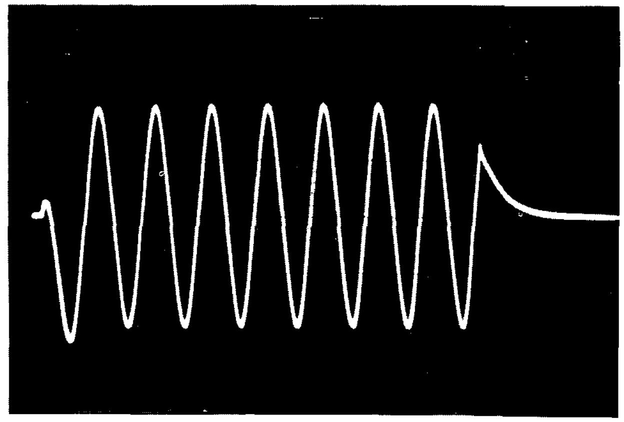

Fig. 10. A portion of the first half-cycle is lost when the system detects the leading edge of a 1-kHz tone burst.

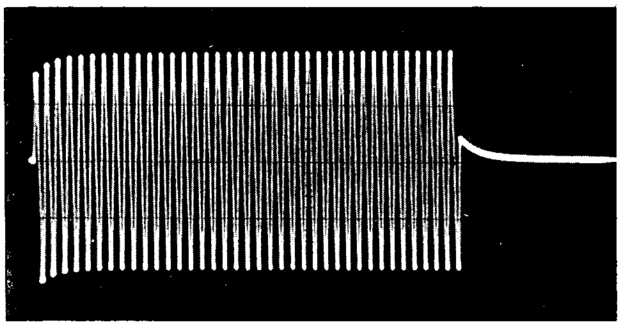

Fig. 11. The 5-kHz tone burst response is affected by the low-pass filtering and phase shift.

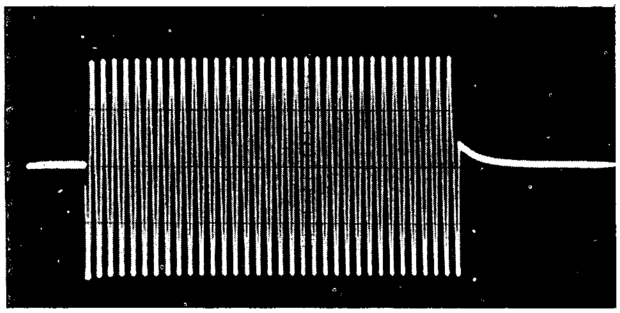

Fig. 12. 1 ms of the leading edge is lost when the system detects a 2-V peak-to-peak 5-kHz tone burst.

4. CONCLUSION

The production version of the system is shown in Fig. 13. This system uses the 15-kHz to 50-kHz energy in the vertical signal from the phonograph record to determine both the onset and the termination of the switching time. While the system is capable of accurately restoring the program content of the signal during the switchover time only at frequencies below 200 Hz, the generally short switching times in the range of 80–600 μs minimize damage to the signal. The short switching time has made possible a large improvement in sensitivity to small ticks and groups of ticks without audibly damaging the program.

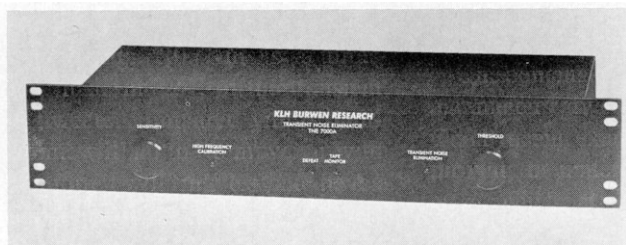

Fig. 13. Model TNE 7000 transient noise eliminator.

THE AUTHOR

Richard S. Burwen received the S.B. and A.M. degrees from Harvard University in 1949 and 1950, respectively.

He was involved in circuit design at Bell Telephone Laboratories, Spencer-Kennedy Laboratories, Krohn-Hite Corporation, and Honeywell, Inc., until 1961. For the past eighteen years Mr. Burwen has been an independent circuit design consultant for over fifty companies in the areas of industrial control, medical electronics, power supplies, space vehicle equipment, television, automotive products, airborne equipment, laboratory instruments, linear integrated circuits, and audio. He holds a number of patents and is the author of numerous papers on audio and analog circuits. At the same time Mr. Burwen was one of the founders of Analog Devices, Inc., and Ohmtec Corporation. He is currently technical consultant to Burwen Research, Inc., and several other companies.