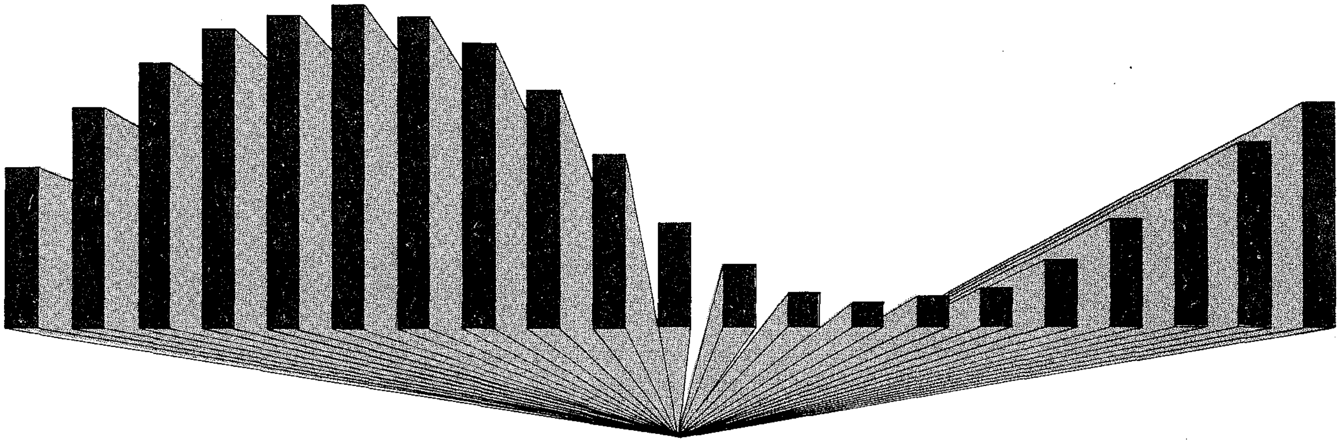

Analog Delay Lines

You can use analog delay lines to produce special audio effects such as echo, reverb, room expansion, and many others. Find out how here.

RAY MARSTON

SOLID-STATE DELAY LINES ARE WIDELY used in modern music and audio systems. They can be used to produce effects such as echo, reverb, chorus, phasing, and flanging (sweepable filtering) for performing artists. In addition, delay lines can produce ambience synthesis and room expansion for home hi-fi systems. Further, a delay line can be used as the heart of a click/scratch filter for a record player, and as a sound-activated switch.

There are two basic types of solid-state delay systems: analog and digital. Digital delay systems tend to be more expensive and more complex than analog types, except when delay times exceed 250 ms, so we'll confine this discussion to analog systems only. But before we get into the hardware, let's talk about psycho-acoustics. In that way we'll learn why the effects produced by delay lines work.

Psycho-acoustics

Many of the special effects obtainable with delay lines depend heavily on how the human brain interprets sounds. The brain does not always perceive sounds as they truly are, but it interprets those sounds so that they conform to a preconceived pattern. Therefore, sometimes the brain can be tricked into misinterpreting those sounds. Scientists have discovered several psycho-acoustic laws that explain how the brain interprets various sounds.

1. If the ears perceive two sounds that are identical in form (waveshape and amplitude) and time-displaced by less than 10 ms, the brain integrates them and perceives them as a single sound.

2. If the ears perceive two sounds that are identical in form but time-displaced by 10–50 ms, the brain perceives them as two independent sounds, but integrates their information content into a single easily recognizable pattern, with no loss of information fidelity.

3. If the ears perceive two sounds that are identical in form but time-displaced by more than 50 ms, the brain perceives them as two independent sounds, but may be unable to integrate them into a recognizable pattern.

4. If the ears perceive two sounds that are identical in shape but not in magnitude, and which are time-displaced by more than 10 ms, the brain interprets them as two sound sources (primary and secondary), and draws conclusions concerning both the location of the primary sound source and the distance between the two sources.

Regarding location identification, the brain identifies the first perceived signal as the primary sound source, even if its magnitude is substantially lower than that of the second perceived signal. That is known as the Hass effect. So a delay line can be used to trick the brain into wrongly identifying the location of a sound source.

Regarding distance identification, the brain correlates distance and time-delay at roughly 0.3 meters (about 13 inches) per millisecond of delay. So a delay line can be used to trick the brain about how far apart two sound sources are.

5. The brain uses echo and reverberation (repeating echoes of diminishing amplitude) information to construct an image of environmental conditions. For example, if a sound reverberates for a total of two seconds, and an echo occurs every 50 ms, the brain may interpret its environment as being a 50-foot cave or a similar hard-faced structure. But if the reverberation time is only 150 ms, the brain may interpret the environment as being a softly-furnished 50-foot room. Those sorts of facts can be used to trick the brain into drawing false conclusions concerning its environment; in fact, it is those tricks that are used in ambience synthesizers and room expanders.

6. The brain is highly sensitive to sudden increases in sound intensity, such as clicks and scratches on record albums, but it is insensitive to sudden decreases in intensity. We're speaking now of transients that last only a few milliseconds. Those facts are what make click/scratch filters possible. We'll discuss how to do that later, but first let's discuss some of the basics of how delay lines work.

Delay-line basics

Modern analog delay lines come in integrated-circuit form and are commonly known as CTD (*C*harge *T*ransfer *D*evice) or bucket-brigade delay lines. Those devices contain a number of analog memory cells (buckets). Each cell is actually a sample-and-hold circuit and, usually, 512, 1024, or 4096 cells are wired in series. Analog input signals are applied at the first cell and the delayed output is taken from the last cell.

Figure 1-*a* illustrates the basic components of an analog delay line. Each bucket

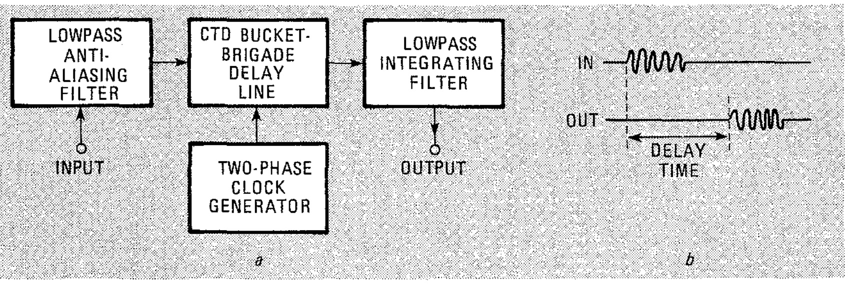

FIG. 1—AN ANALOG DELAY LINE is composed of a bucket-brigade delay line, input and output lowpass filters, and a two-phase clock (*a*). The circuit produces a time-delayed replica of the input signal (*b*).

consists of a small-valued capacitor and a MOSFET which function together as a sample-and-hold stage. As shown in Fig. 1-*b*, a signal fed to the input of a delay line will appear at the output at a later time; as we'll see below, that time depends on both the clock frequency and the number of buckets.

The input signal is applied to the delay line via a lowpass filter that has a cutoff frequency one third (or less) of the clock frequency. The lowpass filter is used to overcome aliasing (intermodulation) problems. The output of the delay line is passed through a second lowpass filter, which also has a cutoff frequency one third (or less) of that of the clock. That filter also rejects clock-breakthrough signals and integrates the output of the delay line into a faithful (but time-delayed) replica of the original input signal.

As shown in Fig. 2, an electronic switch is placed at the input of the delay line; that switch is externally biased to a pre-set voltage. Charges can then be shifted from cell to cell, one cell at a time, at a rate determined by an external two-phase clock. One phase of the clock controls shifting, and the other phase drives the input-sampling switch. The operating sequence is as follows.

On the first half of the clock cycle, the contents of each bucket are shifted to the next bucket in line and a sample of the input signal is fed to the first bucket via S1. On the second half-cycle, all charges (including the most recently sampled one) are shifted to the next bucket, but the input is not sampled. That double-shifting process repeats on each clock cycle, and input samples are repeatedly taken and then shifted.

Near the end of the delay line two sections of buckets are wired in parallel. Those sections differ in that one section has an extra bucket. There is also a phase difference between the clock signals applied to the parallel sections. So the circuit has two outputs which, when added together, effectively fill in the "gaps" in the main delay line. The outputs can be summed either by shorting them together or, preferably, by connecting them to a BALANCE potentiometer, as shown in Fig. 2. The final output of the delay line is thus a time-delayed replica of the original input signal.

Figure 3 shows the essential components of an analog delay circuit. The delay-line MOSFET's have a tetrode structure, so the IC needs two supply voltages (V_{DD} and V_B), plus a ground connection. The input terminal must be biased into the linear mode by resistors R1 and R2. The two outputs of the device are added together, as discussed earlier. Last, the IC must be provided with a two-phase clock signal; usually that signal is a pair of out-of-phase squarewaves that switch fully between V_{DD} and ground. Some analog delay lines have clock dividers so that only a single-phase clock input is needed. We'll discuss specific IC's later in this article.

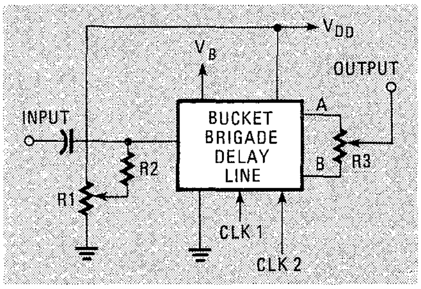

FIG. 3—A TWO-PHASE CLOCK and several biasing resistors are required to drive a bucket-brigade delay line.

How much delay?

The cells of an analog delay line are alternately empty and charged. Each complete clock cycle shifts a charge two stages along the delay line. Therefore the maximum number of samples that a line can take is equal to half the number of stages. For example, a 1024-stage line can take only 512 samples. The actual time delay available from a line is given by $t = ST/2$, where t is the time in seconds, S is the number of stages, and T is the period of the clock. In terms of clock frequency, that equation is $t = S/2f$.

For example, a 1024-stage delay line using a 10-kHz (100 μs) clock gives a delay of $1024 / (2 \cdot 10{,}000) = 51.2$ ms. A 4096-stage line gives a 204.8-ms delay at the same clock frequency.

It's worth pointing out that the bandwidth of an analog delay line is limited to about one third of the clock frequency. Hence a delay line clocked at 10 kHz has a useful bandwidth of only about 3.3 kHz.

Simple effects

Figures 4–15 illustrate a variety of applications for analog delay lines. In these block diagrams we will, for the sake of simplicity, not show the usual input and output lowpass filters. An actual circuit

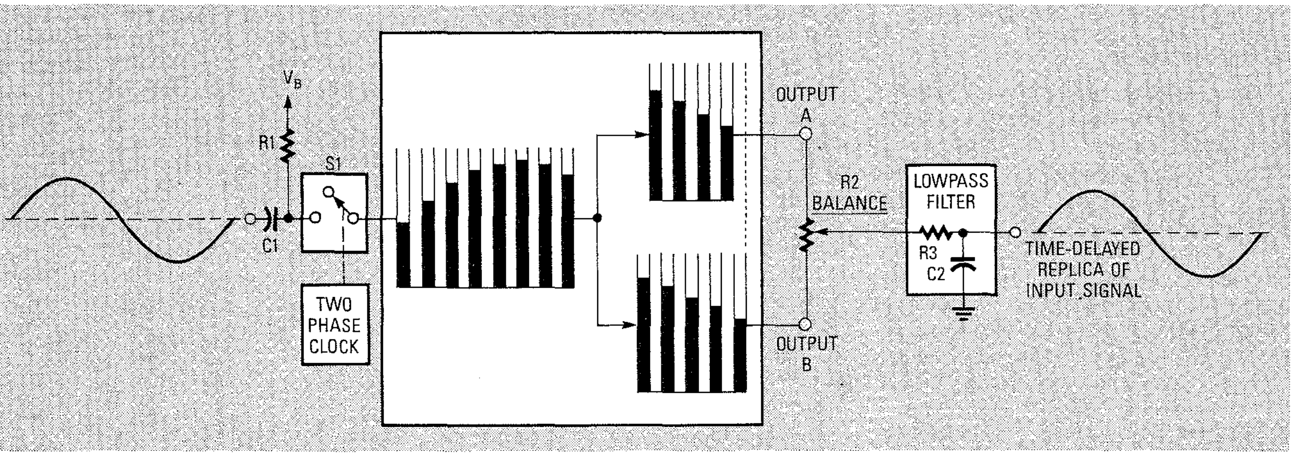

FIG. 2—THE BUCKET-BRIGADE DELAY LINE operates by sequentially shifting charges through alternate cells. The two outputs are summed and filtered for further processing.

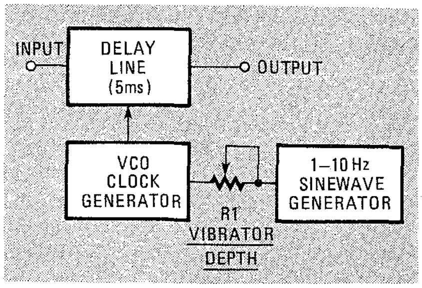

FIG. 4—THE VIBRATO CIRCUIT frequency modulates the input signal with a low-frequency oscillator.

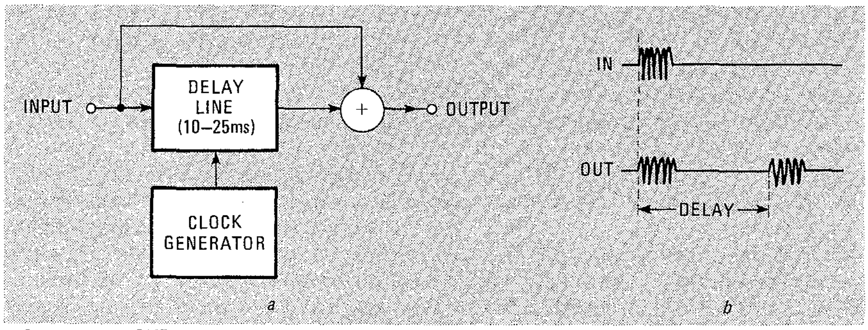

FIG. 5—THE DOUBLE-TRACKING CIRCUIT, or micro chorus, mixes the input signal with a delayed replica of the input signal (*a*). Input and output waveforms are shown in *b*.

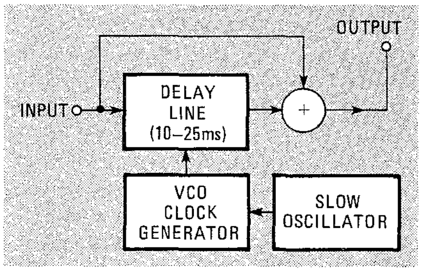

FIG. 6—THE AUTOMATIC DOUBLE-TRACKING, or mini-chorus, frequency modulates the delayed signal.

would have to include filters with the appropriate cutoff frequencies.

Figure 4 shows how the delay line can be used to apply *vibrato* (frequency modulation) to a signal. The low-frequency sinewave generator modulates the clock generator's signal and thereby causes the output signal of the delay line to be time-delay modulated.

Figure 5-*a* shows a delay line used to give a double-tracking effect. The delay is in the perceptible range (10–25 ms), and the delayed and direct signals are added in an audio mixer to give the composite output. Figure 5-*b* illustrates the circuit's input and output signals. One use for a double-tracking circuit is to feed a solo singer's voice through the unit. The output sounds as if two performers were singing in very close harmony. The double-tracking circuit is also called a mini-echo and a micro-chorus.

Figure 6 shows how the circuit shown in Fig. 5 can be modified to act as an *A*uto *D*ouble *T*racking (ADT) circuit. Such a circuit is also called a mini-chorus. Clock signals are derived from a VCO that is modulated by a slow oscillator, so that the delay times slowly vary. The effect that results is that a solo singer's voice sounds like a pair of singers in loose or natural harmony.

Figure 7 shows how three ADT circuits can be wired together to form a "chorus" machine. Each delay line has a slightly different delay time. The original input and the three delay signals are summed; the net effect is that a solo singer sounds like a quartet, or a duet sounds like an octet, etc.

Comb filters

Figure 8 shows a delay line configured as a comb filter. In that circuit, the direct and the delayed signals are added together. In-phase components of both signals increase output amplitude, and those that are out of phase cancel each other and reduce output level. Consequently, as shown in Fig. 8-*b*, the frequency response shows a series of notches; the spacing between notches is the reciprocal of the delay-line time. In the example circuit, the 1-ms delay gives a 1-kHz (1/0.001) notch spacing. Those phase-induced notches are typically about 20–30 dB deep.

The two most popular musical applications of the comb filter are in phasers and flangers. In the phaser (shown in Fig. 9) the notches are swept slowly up and down the audio band via a slow-scan oscillator. Selected components of the input signal are then effectively deleted from the output signal.

The flanger circuit (shown in Fig. 10) differs from the phaser in that the mixer is placed ahead of the delay line, and part of the delayed signal is fed back to one input of the mixer, so that the in-phase signals

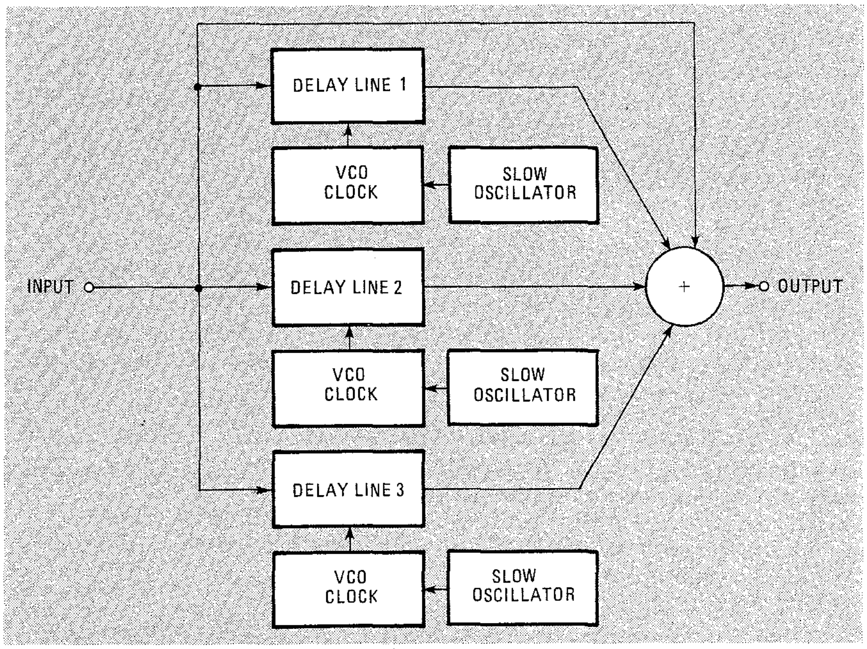

FIG. 7—A CHORUS GENERATOR is composed of several double-tracking circuits.

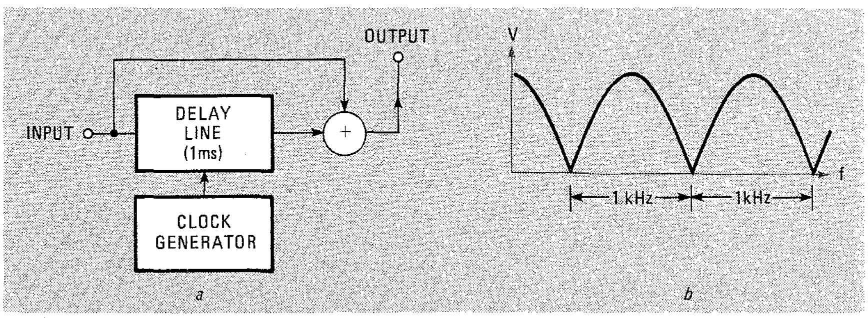

FIG. 8—A COMB FILTER can be built from a delay line, a clock, and a mixer (*a*). With a 1-ms delay, the notches are about 20–30 dB deep, and 1 kHz apart (*b*).

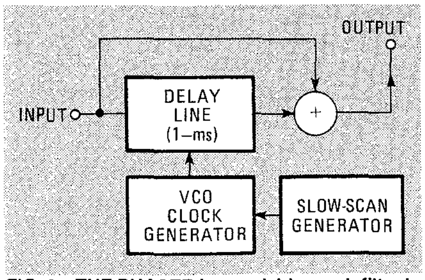

FIG. 9—THE PHASER is a variable comb filter in which the notches are slowly swept up and down the audio band.

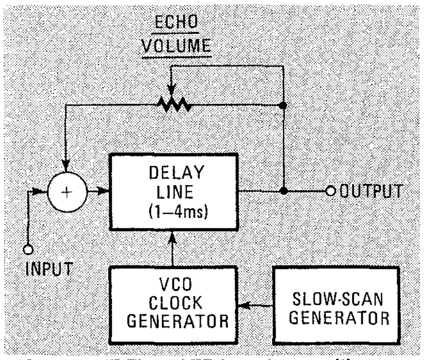

FIG. 10—THE FLANGER is a phaser with accentuated and variable notch depth.

add together regeneratively. Peak amplitudes depend on the amount of feedback, and they can be made very steep. Those phase-induced peaks introduce very powerful acoustic effets as they are swept up and down the audio band.

Echo and reverb circuits

Figure 11 shows how to build an echo circuit with a delay line. The delay (echo) may vary from 10 to 250 ms, and it is adjustable, as is the amplitude of the echo. Note that this circuit produces only a single echo, as shown in Fig. 11-*b*.

The echo/reverb circuit of Fig 12 produces multiple echoes, or reverberation. It uses two mixers, one before the delay line and one after it. Part of the delayed output is fed back to the input mixer, and that is what allows the circuit to give echoes of echoes, echoes of echoes of echoes, and so on. As shown in Fig. 12-*b*, the amplitude of each echo is less than that of its predecessor.

Reverb time is defined as the time it takes for the repeating echo to fall by 60 dB relative to the original input signal; reverb time depends on the delay time and the overall attenuation of the feedback signals. Echo delay time, echo volume, and reverb time are all independently variable.

Hi-fi effects

Figure 13 shows a block diagram of an ambience synthesizer or room expander. The outputs of a stereo system are summed to provide a mono image; the resulting signal is then passed to a pair of semi-independent reverb units (which produce repeating echoes, but not the original signal). The reverb outputs are then summed and passed to a mono amplifier and speaker; the latter is usually placed behind the listener. The system effectively synthesizes the echo and reverb characteristics of a chamber of any desired size, depending on control settings, so the listener can be given the impression that he is sitting in a cathedral, a concert hall, a small club, etc., although in fact he is sitting in his own living room.

There are many possible variations on that circuit. For example, the mono signal may be derived by taking the difference between (rather than the sum of) the stereo signals. Doing that cancels center-stage signals and overcomes a rather disconcerting "announcer-in-a-cave" effect that occurs in "summing" systems.

Predictive switching circuits

Delay lines are particularly useful in solving "predictive" or "anticipatory" switching problems, in which a switching action is required to occur slightly *before* a random event occurs.

Suppose, for example, that you need to record random or intermittent sounds (thunder, speech, etc.). It would be inefficient and expensive to run a tape recorder continuously, but it would be impractical to try to activate the recorder via a sound switch, because part of the sound will already have passed by the time the recorder's motors get up to speed.

Figure 14 shows a delay-line solution to that problem. The sound input activates a switch that (because of mechanical inertia) turns the recorder's motor on in about 20 ms. In the meantime, the sound travels through the 50-ms delay line towards the recorder's audio input. When the sound arrives, the motors have already been turning for 30 ms, so no sound is lost. When the original sound ceases, the sound switch turns off, but the switch extender keeps the motor running for another 100 ms or so, and that enables the entire delayed signal to be recorded.

Click/scratch filter

To conclude our discussion of applications, Fig. 15 shows how predictive switching can be used to help eliminate the sounds of clicks and scratches from an audio system. Sounds of that sort can be detected easily by using stereo phase-comparison techniques.

In Fig. 15, signals from a record album

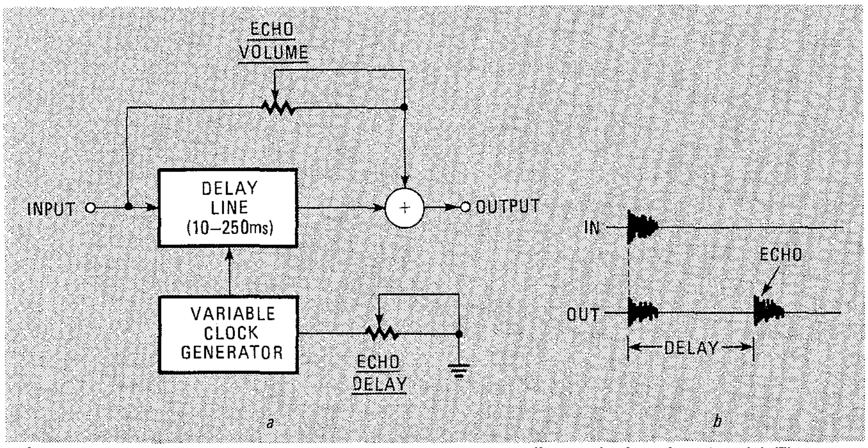

FIG. 11—A VARIABLE ECHO circuit can be built from a delay line, a clock and a mixer (*a*). The echo is actually a time-delayed replica of the input signal (*b*).

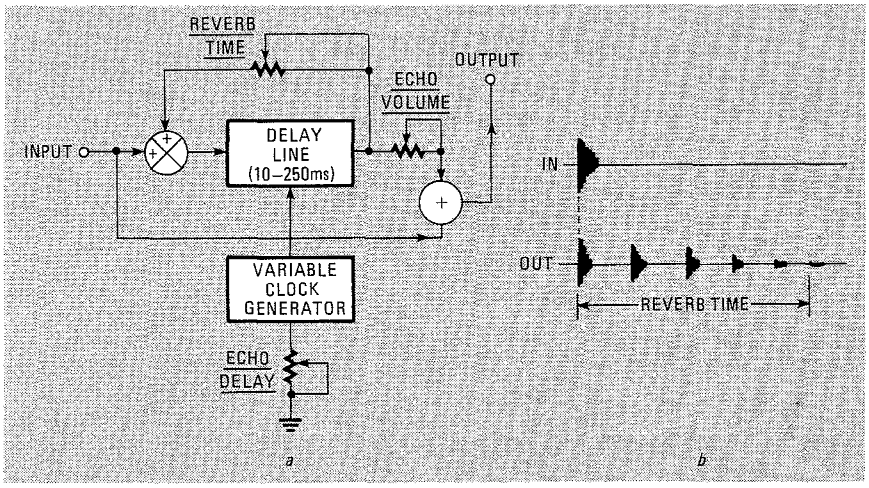

FIG. 12—ADDING REVERB to the echo circuit requires an additional mixer (*a*) to provide progressively diminishing echoes (*b*).

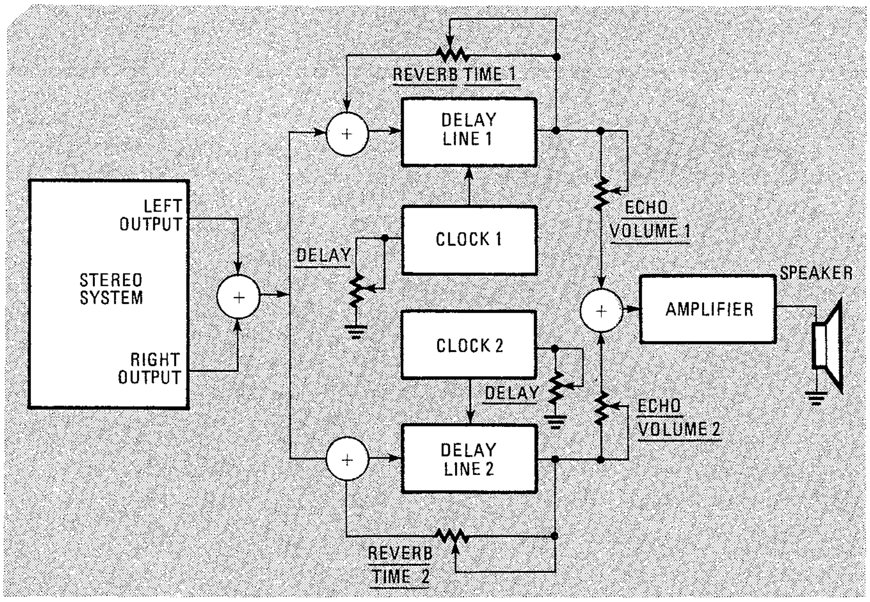

FIG. 13—THE ROOM EXPANDER produces a monophonic delayed output of the sum of the stereo outputs.

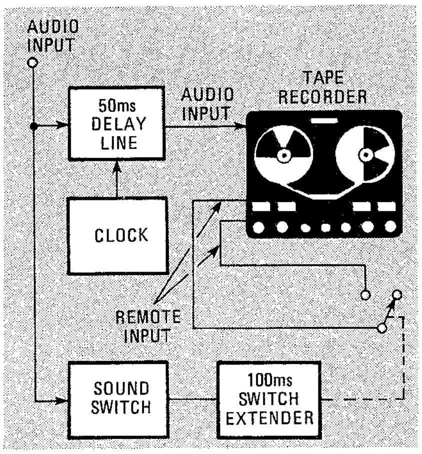

FIG. 14—A SOUND-ACTIVATED RECORDER that does not miss sound input can be built using an analog delay line.

are fed to the audio amplifier via a 3-ms delay line, an analog switch, and a sample-and-hold circuit. Normally the switch is closed, so the signal reaching the audio amplifier is a delayed but otherwise unmodified replica of the input signal. However, when a click or a scratch occurs, the detector/extender circuit opens the switch for 3 ms, and that momentarily blanks the audio signal. Since the ear is relatively insensitive to dropouts, the deletion is not noticed. As you can see in Fig. 15-*b*, when the click in the upper waveform is deleted, the signal is much smoother, as shown in the lower waveform.

Delay-line circuits

Figure 16 shows a practical delay line using an RD5106 (512 stages), RD5107 (1024) stages), or RD5108 (4096 stages) delay line. Those IC's are manufactured by Reticon, Inc. (345 Potrero Avenue, Sunnyvale, CA 94086-9930); Radio Shack should have them in distribution by the time you read this.

Potentiometer R2 should be adjusted so

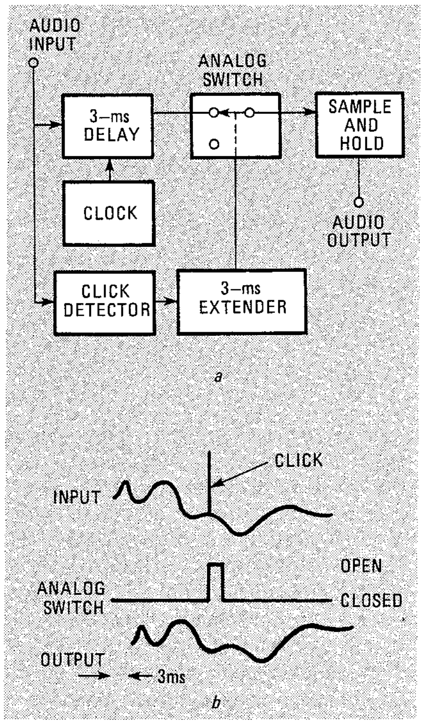

FIG. 15—AN AUDIO CLICK/SCRATCH FILTER can be built with an analog delay line and other components (*a*). The Click Detector opens the Analog Switch for 3 ms to delete a click (*b*).

that symmetrical clipping occurs under over-drive conditions. The maximum clock frequency is 1.6 MHz; the clock should be a fairly sharp squarewave with a risetime of less than 50 ns. The input signal should be limited to 1.5-volts rms to keep distortion less than 1%.

Clock circuits

The clock signal that is fed to a CTD delay line should be reasonably symmetrical, should have fairly fast rise and fall times, and should switch fully between the supply-rail voltages. CMOS devices make good clock generators

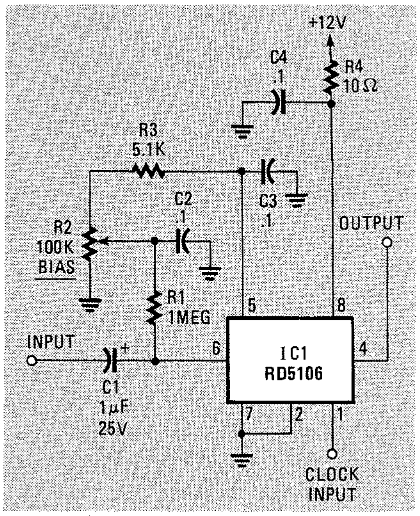

FIG. 16—AN ANALOG DELAY LINE can be built from just a few discrete components. Limit the input signal to less than 1.5-volt rms to keep harmonic distortion less than 1%.

because they are capable of generating waveforms that meet those requirements.

The general-purpose two-phase clock circuit shown in Fig. 17 is inexpensive and can be used in many applications where a fixed or a manually-variable frequency is needed. The frequency can be swept over a 100:1 range via R3; the center frequency can be altered by changing the value of C1.

A high-performance two-phase clock generator based on the VCO section of a 4046 phase-locked loop is shown in Fig. 18. That circuit is useful in applications where the frequency needs to be swept over a very wide range, or needs to be voltage-controlled. The frequency is controlled by the voltage at pin 9. Higher voltages correspond to higher frequencies. Maximum frequency is set by the values of C2 and R1, and minimum frequency is set by the value of C2 and the series values of R3 and R4.

The preceding clock circuits can be used to clock many delay lines. Some, however, have high-capacitance (0.001 μF) inputs and therefore require a low-impedance clock drive. One way of providing a low-impedance clock driver is by connecting two 4013 "D" flip-flops in parallel, as shown in Fig. 19. That circuit is driven by a single-phase clock signal, which can be obtained from either of the preceding circuits.

Filter circuits

As we mentioned previously, in most applications a lowpass filter must be inser-

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DELAY LINES

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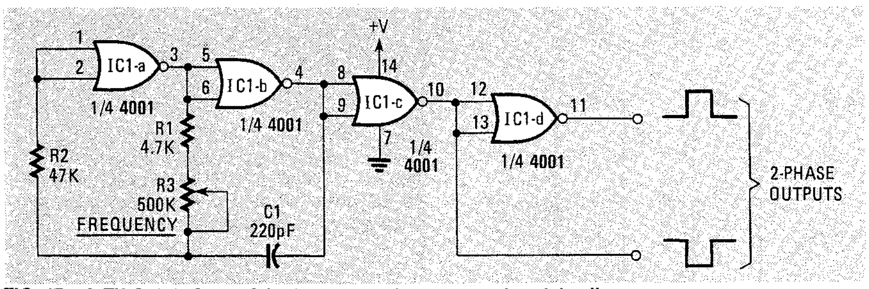

FIG. 17—A TWO-PHASE CLOCK is required by most analog delay lines.

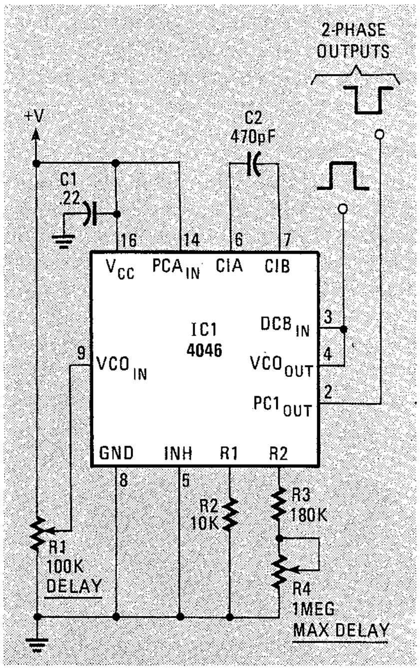

FIG. 18—A VCO CLOCK provides wider range and more precise control over clock frequency.

ted between the actual input signal and the input of the delay line to prevent aliasing problems. In addition, another delay line must be connected in series with the output of the delay line to provide clock-signal rejection and integration of the sample signals. For maximum bandwidth, both filters should have cutoff frequencies that are one third (or less) of the maximum clock frequency. The input filter usually has at least a 1st-order (6-dB per octave) response, and the output filter usually has at least a 2nd-order (12-dB per octave) response.

Figure 20 shows the schematic of a 25-kHz 2nd-order lowpass filter with AC-coupled input and output. The non-inverting terminal of the op-amp (pin 3) is biased at $V_{CC}/2$, usually by a simple resistor network. The cutoff frequency can be varied by giving C2 and C3 alternative values, but they should be maintained in a 1:6 ratio. For example, cutoff frequency can be reduced to 12.5 kHz by giving C2 and C3 values of 0.001 and 0.006 μF, respectively.

All delay lines suffer from a certain amount of insertion loss. Typically, if 100 mV is fed to a delay line, only 70 mV or so will appear at its output. So the lowpass filter on the output should be given some gain, in order to provide an overall circuit gain of one. That is done by designing the filter around an active element, such as an op-amp. **R-E**

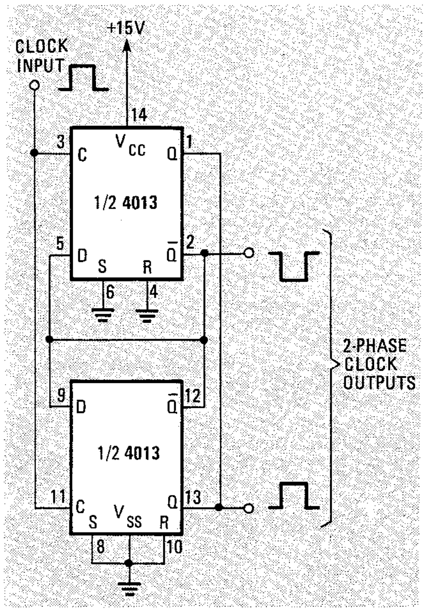

FIG. 19—A LOW-IMPEDANCE single- to two-phase clock converter is used with delay lines having high input capacitance.

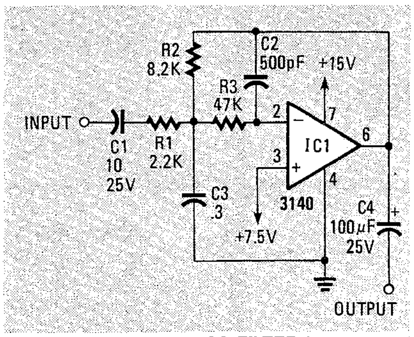

FIG. 20—THIS LOW-PASS FILTER has a cutoff frequency of 25 kHz and 2nd-order response.

RAID ON HBO

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receiver systems require that the RF level at the uplink receiver be kept within certain limits. HBO could, of course, increase power by a factor of 100, and place a 20-dB attenuator at the input to the receiver. But aside from being a crude solution, there's always the problem of getting to the uplink receiver.

Another approach would be to adopt a tactic used by 2-way radio repeaters to avoid unauthorized access. That is to require authorized users to place a subaudible tone on their carriers. The repeater cannot be activated by a signal that lacks the tone. Even more security is provided by using a digital access code.

However, once the transponder has been accessed, there is nothing to stop another signal from taking over the transponder, even if it is not encoded. That signal will control the transponder until the appropriate security circuitry detects an improper or absent access tone or code, causing the transponder to shut down. Therefore, even with such a system a pirate would be able to wreak havoc with the authorized user's signal. And, of course, the problem of reaching existing satellite equipment to make the required modifications remains.

What about the possibility of catching a video pirate? Typical dish beamwidths are 2° to 3°. Because of that, it is virtually impossible to detect a transmission from the ground using conventional means.

How about finding the signal from the air? Assuming that the pirate is using a really lousy dish, that is, one with a 3° beamwidth, sidelobes should be detectable out to about 10°. Sparing you the geometry, that translates to a detection area of about 3.14 square miles. (With a better dish, which is likely, the detection area would be even smaller). Consider for a moment the number of aircraft that would be required to cover the continental United States. And what if the signal originated off shore? Further, those aircraft would have to be in place constantly because no one aside from the pirate knows when he will strike next.

In summary, unless someone turns the pirate in, or unless his ego gets the better of him, the chances of finding Captain Midnight are just slightly higher than zero.

The possibility of deliberate interference to a satellite is a weak link in the satellite-communications chain. That fact has been obvious to the military since the dawn of the satellite age. But the military has the methods, and the money, to make their communications links secure. HBO and other programmers do not. It is little wonder that the industry is terrified. **R-E**