Analogue delay lines

Analogue delay lines can be used to produce special audio effects such as echo, reverb, phasing, flanging, room expansion and predictive switching etc.

Ray Marston

SOLID STATE DELAY LINES are widely used in modern music and audio systems. They can be used to produce popular effects such as echo, reverb, chorus, phasing, flanging etc. in music systems, 'rare' effects such as ambience synthesis or 'room expansion' in expensive hi-fi systems and 'predictive' effects such as click/scratch elimination in record players or auto-switching in tape recorders etc.

Two basic types of solid state delay systems are available, analogue and digital. Digital delay systems tend to be more expensive and complex than analogue types, except where delay times are in excess of 250 ms, so we'll confine our present discussion to analogue systems only.

Delay line basics

Modern solid state analogue delay lines come in integrated circuit form and are almost universally known as CCD (Charge Coupled Device) or 'Bucket-Brigade' delay lines. In essence, these devices contain a stack of analogue memory (sample-and-hold) cells or 'buckets' (usually 512, 1024 or 4096), all wired in series. Analogue input signals are applied at the front of the bucket 'chain' and the delayed output is taken from the main's end.

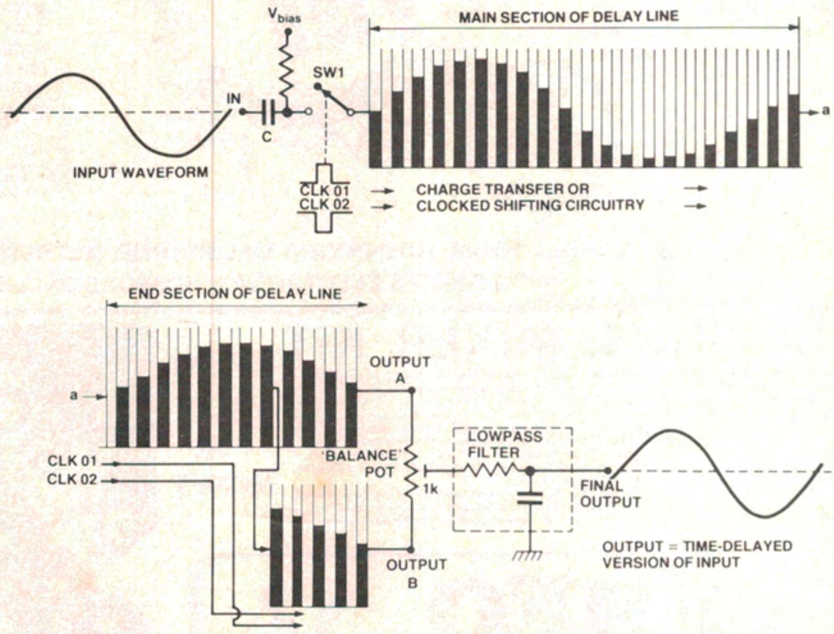

Figure 1. Basic operating principle of the 'bucket brigade' delay line.

Figure 1 illustrates the basic operating principle of an analogue delay line. Each bucket consists of a small capacitor and a tetrode MOSFET and acts like a sample-and-hold stage. An electronic switch is placed at the front of the chain which is externally biased to a preset voltage. Charges can be shifted down the chain, one step at a time, via an external two-phase clock signal; one phase of the clock is also used to activate the input sampling switch. The operating sequence is as follows.

On the first clock half-cycle, each existing bucket charge is shifted backwards one step to the next bucket in the chain and a sample of the instantaneous input signal is fed to the first bucket via SW1, where it is 'stored' as an analogue charge. On the second half-cycle, each existing charge (including the input one) is transferred backwards another step to the next bucket in the chain, but the input is NOT sampled via SW1. There is thus always an 'empty' bucket between each charged bucket in the chain. This double shifting process repeats on each clock cycle, with input samples repeatedly being taken and then clocked towards the back of the chain.

In the final section of the delay line, a short section of buckets is wired in parallel with, and fed from, the main delay line, but has one bucket more than the corresponding section of the main line and is clocked in anti-phase. The IC thus has two outputs which, when added together, effectively fill in the 'gaps' in the main delay line bucket chain. The outputs can be 'added' either by shorting them directly together or preferably, by connecting them to a balance pot as shown in the diagram. The final output of the delay line is thus a quantised but time-delayed replica of the original input signal.

Figure 2 shows the essential 'usage' elements of an analogue delay line chip. The delay line MOSFETs use a tetrode structure, so the IC needs two supply lines (V_{DD} and V_{BB}), plus a ground or common connection. The input terminal must be biased into the linear mode by voltage V_{bias}. The two outputs of the device must be added together, as already described; in Figure 2 we've shown addition by direct-shorting. Finally, the IC must be provided with a two-phase clock

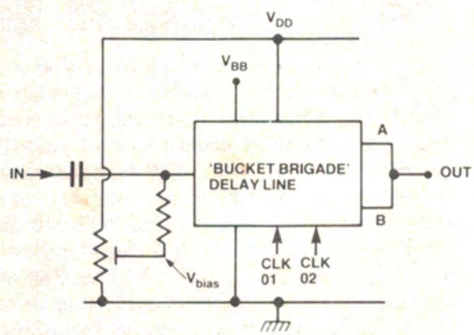

Figure 2. Essential 'usage' elements of an analogue delay line chip.

signal, normally consisting of a pair of anti-phase square waves that switch fully between the V_{DD} and GND (or 0 V) potentials.

How much delay?

We've already seen that the buckets of the analogue delay line are alternately 'empty' and 'charged' and that each complete clock cycle shifts a charge two stages along the bucket chain. Thus, the maximum number of samples taken by a line is equal to half the number of bucket stages (a 1024 stage line can take only 512 samples) and the actual time-delay available from a line is given by:

$$\text{Time Delay} = \frac{S.p}{2} \text{ or } \frac{S}{2.f},$$

where S = number of bucket stages, p = clock period, and f = clock frequency.

Thus, a 1024-stage line using a 10 kHz (100 us) clock gives a delay of 51.2 ms. A 4096-stage line gives a 204.8 ms delay at the same clock frequency. This seems pretty good, but there are two major snags. The first is that the maximum useful signal frequency of the delay line is equal to one third of the clock frequency, so a delay line clocked at 10 kHz has a useful bandwidth of only 3.3 kHz. The second snag concerns costs. Analogue delay lines are rather expensive. A 512-stage device will set you back around $30!

Figure 3 shows the block diagram of a basic, real-life analogue delay line system. The input signal is applied to the input of the delay line via a low-pass filter which has a cut-off frequency that is one third (or less) of the operating frequency of the clock generator, and is used to overcome 'aliasing' or intermodulation problems. The output of the delay line is passed through a second low-pass filter which also has a cut-off frequency one third (or less) of that of the clock. This serves the multiple purposes of rejecting clock break-through signals and integrating the delay line output pulses so that the final analogue output signal is a faithful (but time-delayed) copy of the original input signal.

We'll take a closer look at some of the elements of the Figure 3 circuit and at some practical delay line chips later in this article. In the meantime, let's digress slightly and look at the subject known as psycho-acoustics.

Psycho-acoustics

Many of the special effects that are obtainable with delay lines depend heavily on the human brain's idiosyncratic behaviour when interpreting sounds. Basically, the brain does not always perceive sounds as they truly are, but actually 'interprets' them so that they conform to a pre-conceived pattern. The brain can sometimes be tricked into mis-interpreting the sounds. The study of this particular subject is known as psycho-acoustics. Here are some psycho-acoustic 'laws' that are worth knowing:-

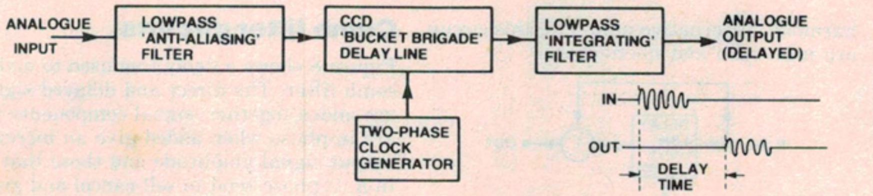

Figure 3. Block diagram of a basic analogue delay line system.

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

(2) If the ears receive 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 recognisable pattern, with no loss of information fidelity.

(3) If the ears receive two signals that are identical in form but time-displaced by greater than 50 ms, the brain perceives them as two independent sounds but may be unable to integrate them into a recognisable pattern.

(4) If the ears receive two sounds that are identical in basic form 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 (a) the location of the primary sound source and (b) the relative distances apart of the two sources.

Regarding 'location' identification, the brain identifies the first perceived signal as the prime sound source, even if its magnitude is substantially lower than that of the second perceived signal (the Hass effect). Delay lines can thus 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 in terms of roughly 0.3 metres per millisecond of delay. Delay lines can thus be used to trick the brain about distance information.

(5) The brain uses echo and reverberation (repeating echoes of diminishing amplitude) information to construct an image of environmental conditions, e.g: if echo times are 50 ms but reverb time is two seconds, the brain may interpret its environment as being a 15 m cave or similar hard-faced structure, but if the reverberation time is only 150 ms it may interpret its environment as being a 15 m wide softly-furnished room. Delay lines can thus be used to trick the brain into drawing false conclusions concerning its environment, as with ambience synthesisers or 'room expanders'.

(6) The brain is highly sensitive to sudden increases in sound intensity (transients of millisecond duration), such as clicks and scratches on discs, but is insensitive to transient decreases in intensity. Delay lines can be used to take advantage of this effect in record players where they can be used (in conjunction with other circuitry) to effectively predict the arrival of a click/scratch and replace it with a neutral or negative transient.

APPLICATIONS

Simple musical effects

Figures 4 to 15 illustrate a variety of analogue delay line applications. In these diagrams we have, for the sake of simplicity, ignored the presence of the usual input/output low-pass filters. Let's start by looking at some simple musical effects circuits.

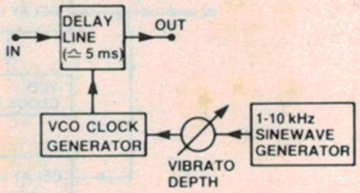

Figure 4. True vibrato circuit which applies slow frequency-modulation to all input signals.

Figure 4 shows how the delay line can be used to apply vibrato (frequency modulation) to any input signal. The low-frequency sinewave generator modulates the clock generator frequency and thus causes the output signals to be similarly time-delay modulated. Simple.

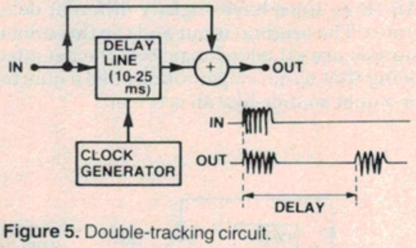

Figure 5. Double-tracking circuit.

Figure 5 shows the delay line used to give a double-tracking effect. The delay time is in the 'perceptible' range 10-25 ms and the delayed and direct signals are added in an audio mixer to give the composite 'two signals' output shown in the diagram. If a solo singer's voice is played through the unit it sounds like a pair of singers in very close ▶

harmony. Alternative names for this circuit are 'mini-echo' and 'micro-chorus'.

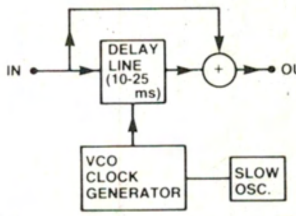

Figure 6. Auto-Double-Tracking (ADT) or mini-chorus circuit.

Figure 6 shows how the above circuit can be modified to act as an Auto-Double-Tracking (ADT) or mini-chorus unit. Clock signals are derived from a VCO that is modulated by a slow oscillator so that the delay times slowly vary. The effect is that when a solo singer's voice is played through the unit it sounds like a pair of singers in loose or natural harmony.

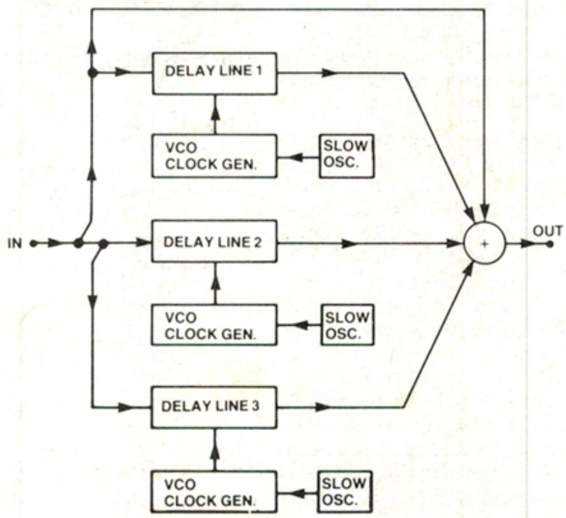

Figure 7. 'Chorus' generator.

Figure 7 shows how three ADT circuits can be wired together to make a 'chorus' machine. All three lines have slightly different delay times. The original input and the three delay signals are all added together, the net effect being that a solo singer sounds like a quartet, or a duet sounds like an octet, etc.

Comb filter circuits

Figure 8 shows a delay line used to make a comb filter. The direct and delayed signals are added together; signal components that are in-phase when added give an increased output signal amplitude and those that are in anti-phase tend to self-cancel and give a reduced output level. Consequently, the frequency response shows a series of notches, the notch spacing being the reciprocal of the line delay time (1 kHz spacing at 1 ms delay, 250 Hz spacing at 4 ms delay).

These phase-induced notches are typically only 20-30 dB deep.

The two most popular musical applications of the comb filter are in 'phasers' and 'flangers'. In the phaser (Figure 9) the notches are simply swept slowly up and down the audio band via a slow-scan oscillator, introducing a pleasant acoustic effect on music signals.

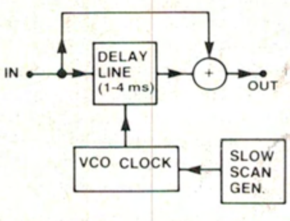

Figure 9. A phaser is a variable comb filter in which the notches are slowly swept up and down the audio band.

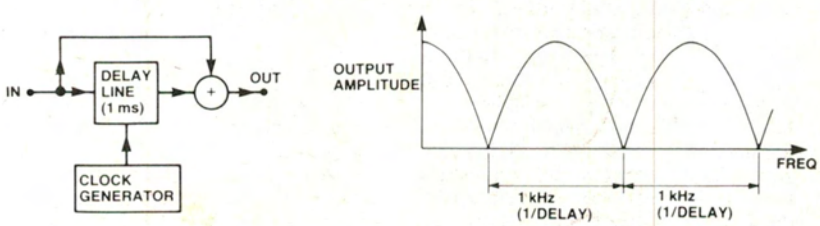

Figure 8. CCD comb filter. Notches are about 20-30 dB deep, 1 kHz apart.

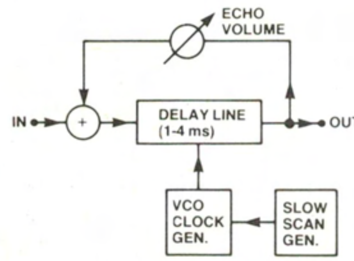

Figure 10. A flanger is a phaser with accentuated and variable notch depth.

The flanger circuit (Figure 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 in-phase signals add together regeneratively. Amplitudes of the peaks depends on the degree of feedback and can be made very steep. These phase-induced peaks introduce very powerful acoustic effects as they are swept up and down through music signals via the slow-scan oscillator.

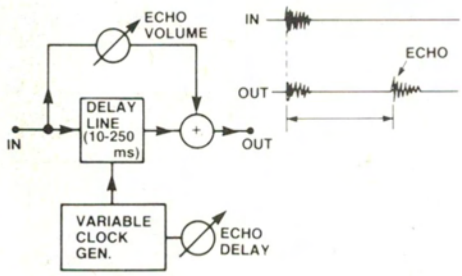

Figure 11. An echo unit.

Echo/reverb circuits

Figure 11 shows the basic circuit of an echo unit. The delay (echo) may vary from 10 ms to 250 ms and is usually adjustable, as is the echo amplitude. Note that this circuit produces only a single echo.

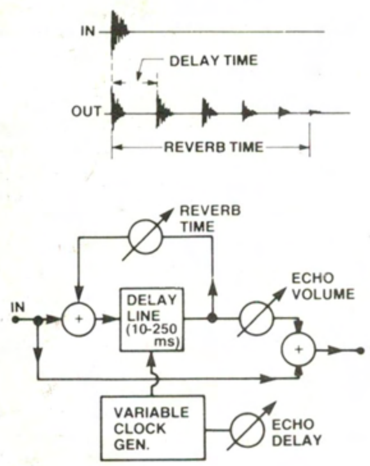

Figure 12. Echo/reverb unit.

The echo/reverb circuit of Figure 12 produces multiple or repeating echoes (reverberation). It uses two mixers, one ahead of the delay line and the other at the output. Part of the delay output is fed back to the input mixer so that the circuit gives echoes of echoes of echoes, etc. The reverb time is defined as the time taken for the repeating echo to fall by 60 dB relative to the original input signal and depends on the delay time and the overall attenuation of the feedback signals. Each delay time, echo volume and reverb time are all independently variable.

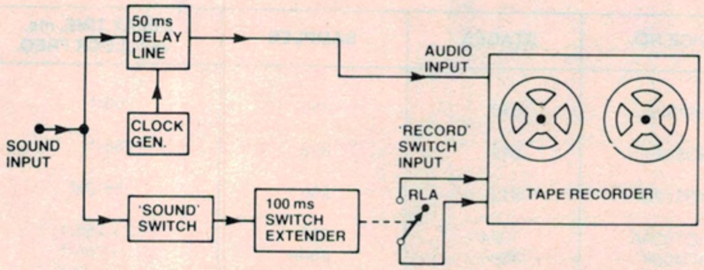

Figure 14. Automatic tape recorder with 'predictive' switching.

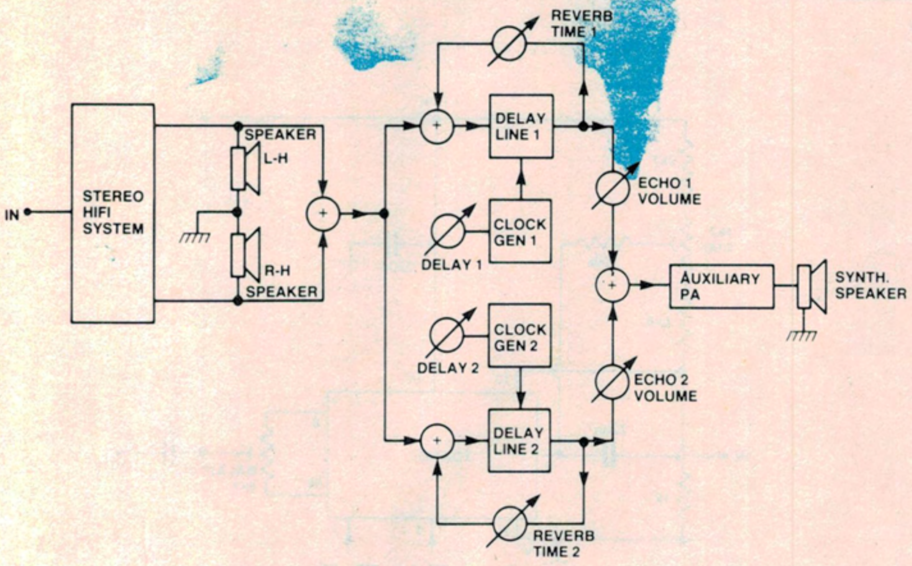

Figure 13. Ambience synthesiser or 'room expander'.

Figure 13 shows the basic circuit of an ambience synthesiser or room expander. Here, the outputs of a conventional stereo hi-fi system are summed to give a mono aural 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 PA system and speaker which is usually placed behind the listener. The system effectively synthesises the echo and reverb characteristics of a chamber of any desired size so that the listener can be given the impression of sitting in a cathedral, concert hall or small club house etc, while in fact sitting in his own living room. Such units produce very impressive results.

There are lots of possible variations on the basic Figure 13 circuit. In some cases the mono signal is derived by differencing (rather than summing) the stereo signals, thereby cancelling centre-stage signals and overcoming a rather disconcerting 'announcer-in-a-cave' effect that occurs in 'summing' systems. The number of delay (reverb) stages may vary from one in the cheapest units to four in the more expensive units.

Predictive switching circuits

Delay lines are particularly useful in helping to solve 'predictive' or 'anticipatory' switching problems in which a switching action is required to occur slightly *before* some random event occurs.

Suppose, for example, that you need to make recordings of random or intermittent sounds (thunder, speech, etc). To have the recorder running continuously would be inefficient and expensive. It would not be practical to try activating the recorder automatically via a sound switch since part of the sound will already have occured by the time the recorder turns on.

Figure 14 shows the solution to this problem. The sound input activates a sound switch which, because of mechanical inertia, turns the recorder's motor on within 20 ms or so. In the meantime, the sound travels through the 50 ms delay line towards the recorder's audio input terminal, so that the recorder has already been turned on for 30 ms by the time the first part of the sound reaches it. When the original sound ceases the sound switch turns off, but the switch extender maintains the motor drive for another 100 ms or so, enabling the entire 'delayed' signal to be recorded.

Finally, to conclude this 'applications' section, Figure 15 shows how 'predictive' switching can be used to help eliminate the sounds of clicks and scratches from a record player. Such sounds can easily be detected by using stereo phase-comparison techniques.

In Figure 15 the disc signals are fed to the audio amplifier via a 3 ms delay line, a bilateral switch and a track-and-hold circuit. Normally, the bilateral switch is closed and the signal reaching the audio amplifier is a delayed but otherwise unmodified replica of the disc signal. When a click or scratch occurs on the disc the detector/extender circuit opens the bilateral switch for a minimum of 3 ms, momentarily blanking the audio signal to the amplifier. Because of the presence of the delay line, the blanking period effectively straddles the 'click' period, enabling its sound effects to be completely eliminated from the system (see 'Psycho Acoustics'). ▶

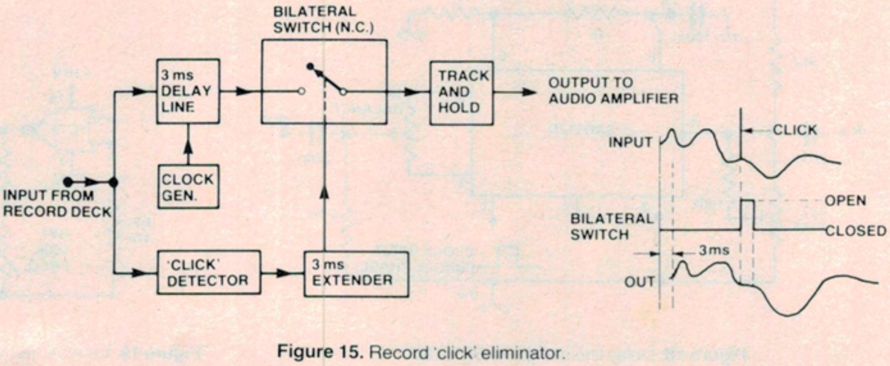

Figure 15. Record 'click' eliminator.

DEVICE NO.	STAGES	SAMPLES	DELAY TIME, ms, VS. CLOCK FREQ.	DELAY AT 7 kHz BANDWIDTH	NOTES
TDA1022	512	256	256/f	12.8 ms	Very popular low-cost delay line
SAD512	512	256	256/f	12.8 ms	512-stage delay line (obsolescent)
SAD512D	512	256	256/2xf	12.8 ms	Built-in clock divider uses single-phase clock
SAD1024A	1024	512	2 x 256/f	25.6 ms	Dual SAD512 delay line
SAD4096	4096	2048	8 x 256/f	102.4 ms	4096-stage delay line. Clock-terminal input Capacitance = 1000 pF

Figure 16. Basic details of five popular CCD delay lines.

PRACTICAL CIRCUITS

Delay lines

Figure 16 shows basic details of five popular CCD delay lines. The TDA1022 and the SAD512 are general-purpose 512-stage delay lines requiring two-phase clock inputs. They give 12.8 ms delay at 7 kHz bandwidth when driven at 20 kHz clock frequency.

The SAD512D is an 'updated' version of the SAD512 and incorporates built-in output drivers and a clock input divider. It requires a single-phase clock input.

The SAD1024A is a dual version of the SAD512. The two halves can be used independently or can be wired in series to give a delay of 25.6 ms at 7 kHz bandwidth.

The SAD4096 gives a performance equal to eight SAD512s in series. It provides a delay of 102.4 ms at 7 kHz bandwidth or 250 ms at 3 kHz bandwidth. The device requires a low-impedance two-phase clock drive, since its clock terminal input capacitance is about 1000 pF.

Figures 17 and 18 show a couple of practical delay line circuits using TDA1022 and SAD512D devices. Both circuits use a preset to adjust the input dc bias so that symmetrical clipping occurs under overdrive conditions and another preset to balance the two outputs for minimum clock breakthrough.

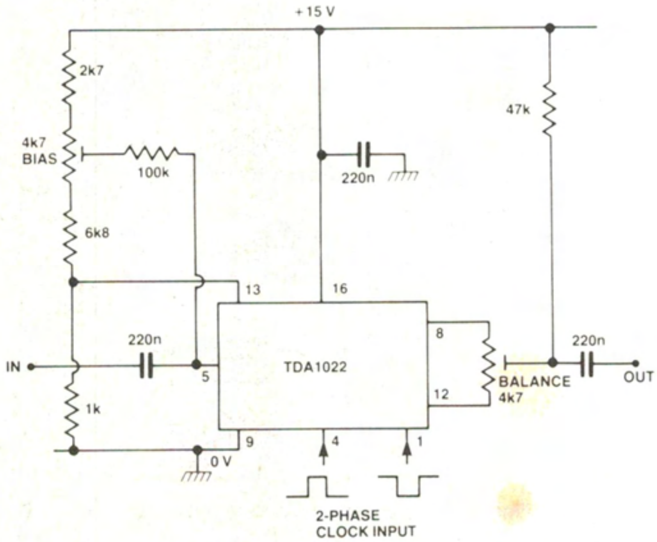

Figure 17. Delay line using the TDA1022.

Clock generators

The clock signals to a CCD 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 ideal clock generators and Figures 19 to 21 show three practical circuits.

The general-purpose two-phase generator of Figure 19 is inexpensive and can be used in most applications where a fixed or manually-variable frequency is needed. The frequency can be swept over a 100:1 range via RV1 and the centre frequency can be altered by changing the C1 value.

The high-performance two-phase generator of Figure 20 is based on the VCO section of a

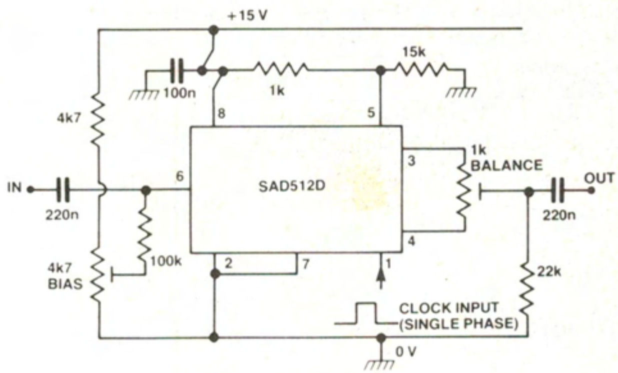

Figure 18. Delay line using the SAD512D.

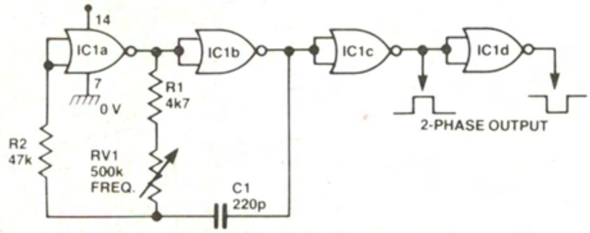

Figure 19. Variable-frequency general-purpose two-phase CMOS clock generator.

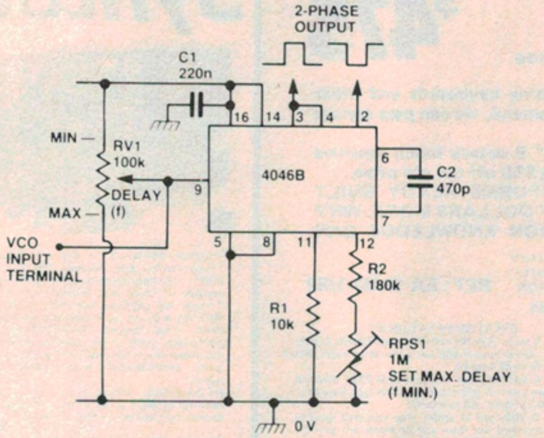

Figure 20. High-performance voltage-controlled two-phase CMOS clock generator.

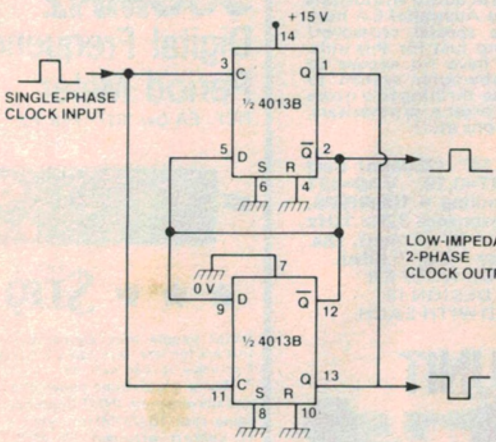

Figure 21. Single-phase to two-phase converter, with low impedance output.

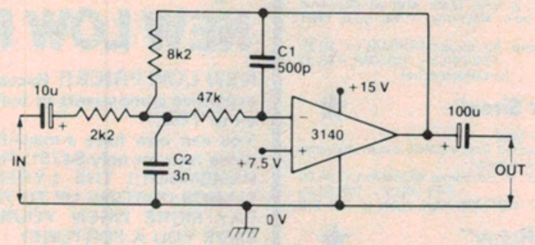

Figure 22. 25 kHz second-order maximally-flat low-pass filter.

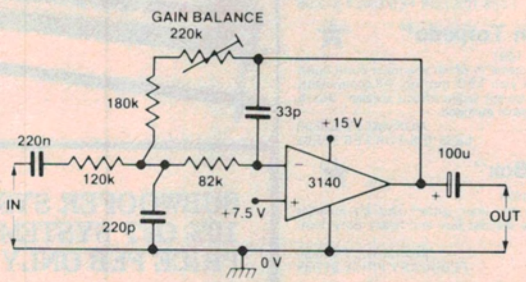

Figure 23. Adjustable-gain second-order low-pass output filter.

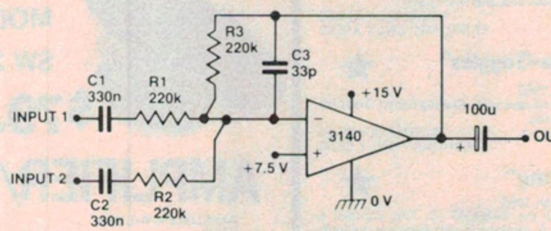

Figure 24. Combined two-input mixer/first-order low-pass filter.

4046B phase-locked loop chip and 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 on pin 9, being at maximum (minimum delay) when pin 9 is high and minimum (maximum delay) when pin 9 is low. Maximum frequency is determined by the C2-R1 value and minimum frequency by the value of C2 and the series values of R2-RPS1.

The Figures 19 and 20 circuits can be used to directly clock all CCD delay lines except the SAD4096, which has a clock terminal capacitance of 1000 pF (1n) and needs low-impedance clock drive. The SAD4096 is best driven by the circuit shown in Figure 21 which uses the two halves of a 4013 divider wired in parallel to give the required low-impedance two-phase output; the circuit is driven by a single-phase clock signal which can be obtained from either of the Figure 19 or 20 circuits.

Filter circuits

In most applications a low-pass filter must be inserted between the actual input signal and the input of the delay line, to prevent aliasing problems. Another must be inserted in series with the output of the line to provide clock-signal rejection and integration of the 'sample' signals. For maximum bandwidth both filters usually have a cut-off frequency that is one third (or less) of the maximum clock frequency used; the input filter usually has a first-order or better response and the output filter has a second-order or better response.

Figure 22 shows the practical circuit of a 25 kHz second-order low-pass filter with ac-coupled input and output. The non-inverting terminal of the op-amp is biased at half-supply volts, usually by a simple potential divider network. The cut-off frequency can be varied by giving C1 and C2 alternative values, but in the same ratio as shown in the diagram, e.g: cut-off can be reduced to 12.5 kHz by giving C1 and C2 values of 1n and 6n respectively.

All delay lines suffer from a certain amount of 'insertion' loss. Typically, if 100 mV is put in at the front of the delay line, only 70 mV or so appears at the output. Often the output low-pass filter is given a degree of compensatory gain to give zero overall signal loss. Figure 23 shows such a circuit. This circuit has a nominal cut-off frequency of about 12 kHz, depending on the setting of the GAIN BALANCE control.

Finally, to complete this look at CCD delay line circuits, Figure 24 shows how a two-input unity-gain 'mixer' (adder) can also be made to act as a first-order low-pass filter by simply wiring a roll-off capacitor (C3) between the output and the terminal of the op-amp. This type of circuit is often used at the front end of CCD flanger and reverberation designs.