BBD SOUND EFFECTS UNIT

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Phasing, vibrato and reverberation are commonly used sound effects in modern music. The effects unit described here is a high-end piece of audio equipment that will make many musicians and sound engineers sit up. Based on a state-of-the-art bucket-brigade delay (BBD) chip, the unit is capable of many popular sound effects, including ADT, chorus, phasing and real-to-life reverberation.

Under normal circumstances, sound travels at a speed of about 340 m per second. This means that short echoes occur in relatively small rooms already, giving the so-called reverberation effect. Acoustic perception experiments have proved that the human ear is capable of detecting a sound delay as small as 5 ms only (corresponding to a distance of about 1.65 m). In particular, short reflections with their associated differences in regard of level, delay, and spectral composition create an impression of space with the listener.

Most electronic musical instruments are not based on sound created in a resonant cavity of any size or shape, and as a result produce a relatively 'flat' sound. Reverberation may be added by electronic means to add warmth to the sound of these instruments. In the present sound effects units, reverberation is achieved with the aid of adjustable degrees of feedback and attenuation, which results in a remarkably natural effect.

Bucket brigade delay

The drawing in Fig. 1 shows the basic set-up of an analogue delay line based on a bucket brigade memory. The memory is essentially a series of sample-and-hold circuits, each of which consists of electronic switches and capacitors. The analogue signals stored in the capacitors are sampled under the control of a central clock signal. At each clock pulse, the sample is shifted one capacitor to the right, hence the name 'bucket brigade' (the precursor of today's fire brigade).

After n clock pulses, the analogue signal is advanced n positions in the memory. A double clock is used to prevent the contents of two 'buckets' affecting each other as a result of the shift operation. Hence, clock 1 and clock 2 are in opposite phase.

To ensure acceptable distortion of the input signal as a result of the sampling operations, the clock frequency must be at least two times the highest frequency to be sampled (Nyquist's sampling theorem). A low-pass filter at the input of the bucket brigade delay limits the frequency range to a usable value. The output of the BBD chip also has a low-pass filter, in this case to remove the clock signal component.

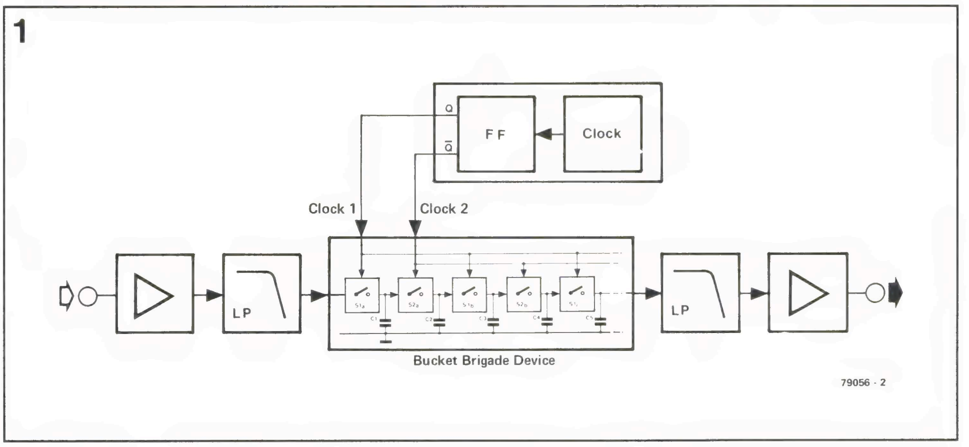

Fig. 1. Block diagram of the bucket-brigade sound effects unit.

Reverberation

Reverberation is an acoustic effect which occurs, in principle, in every room of which the walls have sound reflecting properties. The sound reflections are noted by generating a short acoustic sound burst, e.g., a hand clap. This sound will reach the ear directly as well as indirectly via reflections. The time it takes any reflection to reach the ear is in direct proportion to the time taken by the sound to reach the point where it is reflected. The amplitude of the reflection depends on the length of the path and the acoustic properties of the reflecting surface. Stone walls, for instance, absorb very little sound, whereas curtains have virtually no reflective properties. In many cases, sound reaches the listener via different paths. The amplitude of the reflections as a function of time is illustrated in Fig. 2.

The decay time is the time that lapses before a particular sound is so weak that it is no longer perceived. This parameter depends on the construction of the listening room and the materials used. A natural sounding reverberation effect requires at least 1,000 reflections per second, which, as a further proviso, must reach the listener at a certain irregularity.

Reverberation: the electronic way

Figure 3a shows the simplest configuration of a reverberation unit based on controlled feedback. The associated amplitude-*vs*-time diagram is given in Fig. 3b. To achieve a decay effect, the amplitude of the reflections is reduced as a function of time with the aid of a voltage attenuator. In practice, the reverberation time, *t*, is defined as the time required to reduce the sound energy by a factor of 10^6 (60 dB). In Fig. 3a, this time may be calculated by counting the number of 'needle pulses' between the instant the sound is generated and the instant its amplitude is 60 dB smaller. Next, the number of needles is multiplied with the delay time, τ, of the reverberation unit, hence,

$$t = 60/\alpha\tau$$

where α is the attenuation per passage.

A practical example: a reverberation unit has an attenuation of 3 dB and a delay of 50 ms. An attenuation of 60 dB is, therefore, reached after 20 passages, each of which introduces a delay of 50 ms. The reverberation time, *t*, is 20×50 ms = 1 s.

In practice, reverberation times of the order of one or two seconds pose problems because they require an extensive (i.e., long) delay line. Acceptable results for reverberation times longer than about 0.5 s are only achievable with digital delay lines. Lower attenuation and a greater number of passes are usually not feasible in view of the risk of oscillation.

A second problem arises from the equal distances travelled by the generated reflections. Such a constant pattern can only occur in a spherical room, which, in the case of the above example (50-ms delay time) has a radius of 16.6 m. Evidently, such a room is at best rare in the real world.

An annoying side-effect of equally long t reflections is the creation of a comb-filter response—see Fig. 3c. At a delay of 20 ms, the distance between two peaks equals 100 Hz. The comb filter effect introduces a variation in the attenuation which is simple to calculate if the normal attenuation of the circuit is known. From the above example, a normal attenuation of 3 dB means that the signal is attenuated by a factor of 0.7 after a single passage through the reverberation unit. The variation in the attenuation owing to the comb filter effect is (1+0.7)/(1−0.7), i.e., 5.7 times or about 15 dB. Obviously, an amplitude ripple of 15 dB is not acceptable for hi-fi stereo applications. Yet, many reverberation units produced in the past

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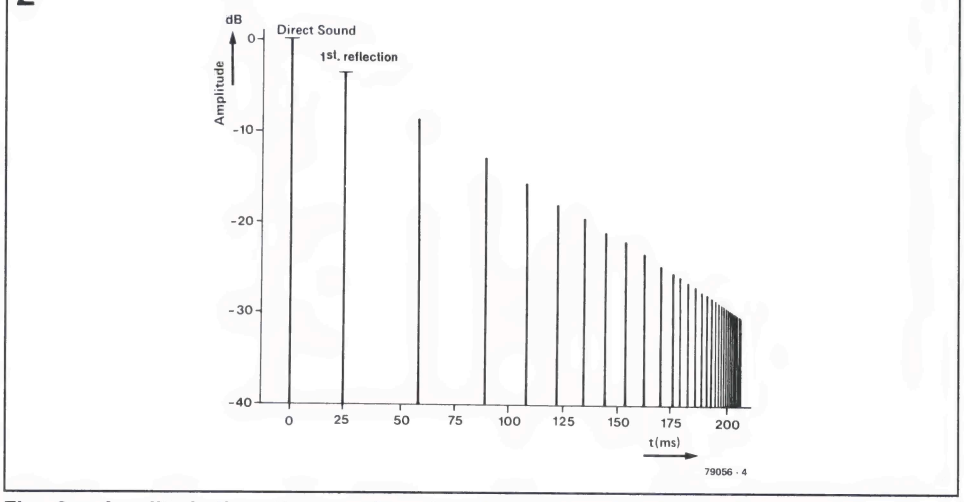

Fig. 2. Amplitude-time diagram of a pulse-shaped sound and the reflections caused by it.

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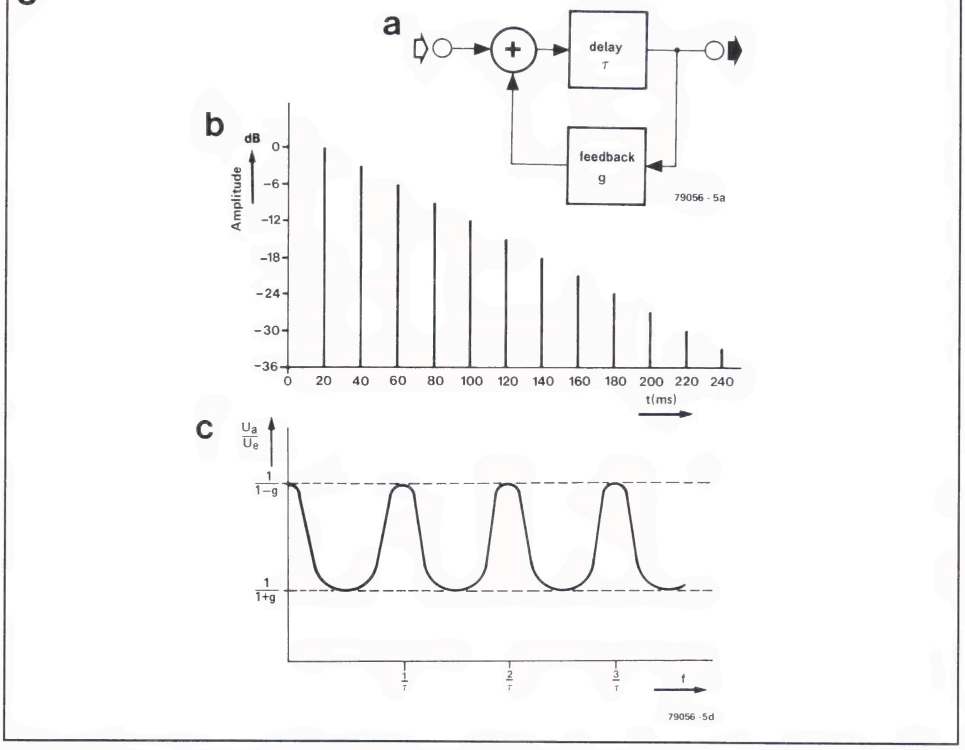

Fig. 3. Simplest configuration of a delay line with feedback (3a) and the associated amplitude-time diagram (3b). The frequency response of such a delay unit is not unlike that of a comb filter (3c).

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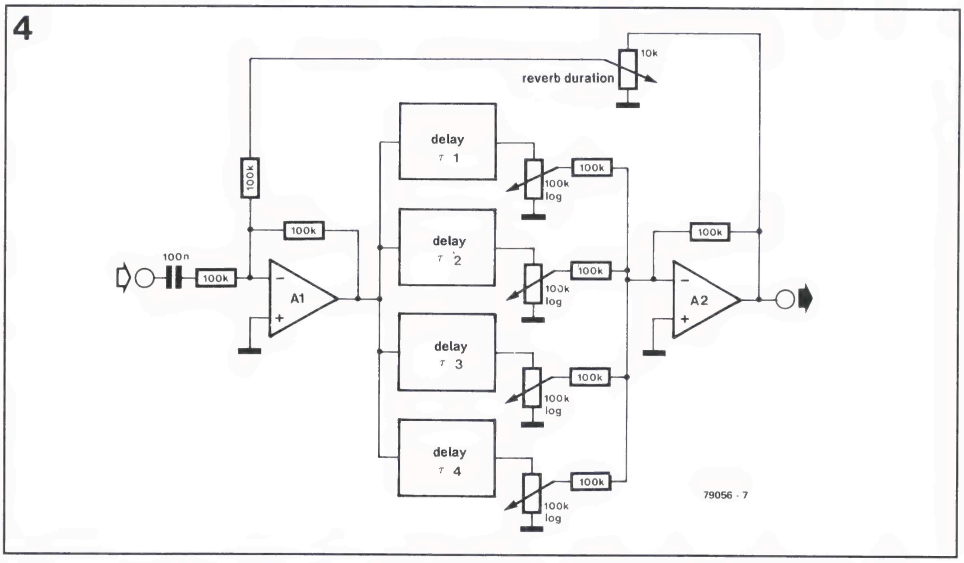

Fig. 4. Improved reverberation principle based on individually controlled delay lines.

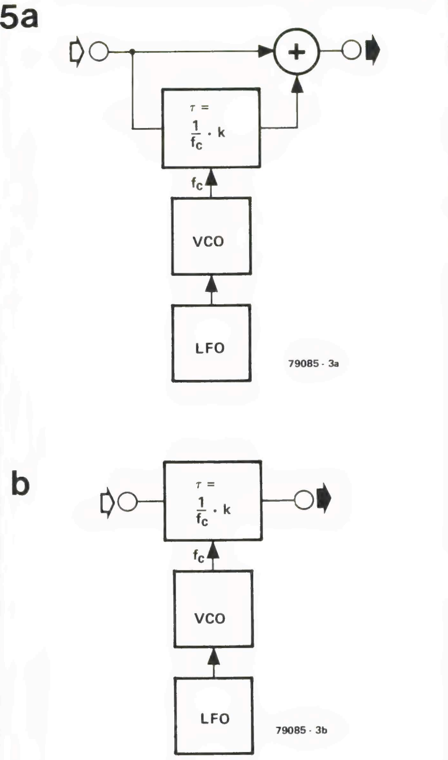

Fig. 5. Basic circuits for phasing (5a) and vibrato (5b).

used the above principle simply for lack of a better (electronic) alternative.

Multiple reverberation

A naturally sounding reverberation effect can only be achieved by using different delays of non-related durations. The block diagram of a reverberation unit based on this principle is shown in Fig. 4. From practical experiments, at least four delays are required for acceptable results. The sound effects unit described here has six different and non-related delays, while the attenuation for each of these is adjustable to give an optimum room simulation.

The delays used in the sound effects unit allow an estimate to be made of the size of the simulated room. A sound delay of 10 ms corresponds to a total path length of 3.3 m, or a wall distance of 1.65 m. The maximum setting, 100 ms, simulates a wall distance of 16.5 m.

Sound effects

The BBD sound effects unit offers a variety of sound effects which may be set to individual liking with a large number of controls. As examples, the degree of feedback for the delayed signals may be adjusted; the 'clean' (input) signal may be mixed with any one delayed signal. Furthermore, the unit allows a single delay to be used.

The **ADT-effect** (automatic double-tracking) is commonly used in modern music technology to give the sound more substance. Basically, the signal is briefly delayed (τ=1–5 ms) and then mixed with the original. If used in a multiple way, the result is the **Chorus-effect**. Here, the delay time is not constant but subject to small, irregular changes caused by modulation of the clock signal by a pseudo-random signal generator, for which the sound effects unit has an external input. The chorus-effect may use one or more delay lines in the unit, provided the output signal is not fed back to the input. The delayed signal is, therefore, simply added to the output signal.

Vibrato and **phasing** are based on modulation of the clock signal with a triangular, low-frequency, signal supplied by, for instance, an LFO (low-frequency oscillator). The vibrato-effect is obtained by using the delayed signal only, while for phasing the modulated as well as the delayed signal are added to the output signal. The different ways of generating these two sound effects are illustrated in Figs. 5a and 5b. The sound effects are rather different also. Strong vibrato brings to the mind a worn tape recorder or gramophone with speed regulation problems, while phasing is associated by many with the Hammond-effect based on doppler shift and achieved with the aid of rotating loudspeakers.

Phasing uses the previously mentioned comb-filter response that occurs at short delays (refer back to Fig. 3). The modulation of the clock signal shifts the points of maximum attenuation (poles) of the comb filter (Fig. 6) periodically, and so causes a spatial sound effect.

Both vibrato and phasing use delays smaller than 10 ms, which allows ready use of a BBD IC.

MN3011/MN3101 BBD chip set

The Type MN3011 from Panasonic (a Matsushita company) contains a 3328-stage bucket-brigade delay line in PMOS technology. The pin-out and internal configuration diagram in Fig. 7 shows that six taps on the delay are bonded out to pins. The delay times associated with these pins are suitable for reverberation applications. The shortest reverberation time is available at pin 9, the longest at pin 4. The actual delay times achieved with the IC are determined by the frequency of the clock signals which are applied in opposite phase to pins 2 and 10. The maximum and minimum delay times at two clock frequencies, 10 kHz and 100 kHz, are given below:

BBD output (pin)	Delay at f_{cl}=10 kHz (ms)	Delay at f_{cl}=100 kHz (ms)
1(9)	19.8	1.98
2(8)	33.1	3.31
3(7)	59.7	5.97
4(6)	86.3	8.63
5(5)	139.5	13.95
6(4)	166.4	16.64

The operating voltage of the MN3011 is 15 V typical and 18 V maximum. At a supply voltage of 15 V, the current consumption is 8 mA typical. The direct voltage at the signal input must be adjusted for minimum distortion. Starting from a level of half the supply voltage, a potential change of up to 2 V may be necessary. The amplification of the BBD chip is 0 dB typical (unity gain), but may lie within 4 dB of this value. The maximum input level is stated as 1V_{rms} at 2.5% transient harmonic distortion (THD). At the nominal input level of 770 mV_{rms} (1 kHz), the THD is 0.4% typical.

A bandwidth of 10 kHz is achievable at a clock frequency of 40 kHz. From practical measurements on the chip, the noise level is between –70 and –72 dB at a clock of 40 kHz.

The pinning and internal configuration of the clock driver chip Type MN3101 are given in Fig. 8. External components are used to determine the clock oscillator frequency, which is divided by two and subsequently shaped to provide the opposite-phase clock signals for the MN3011. The MN3101 has an on-board voltage source that supplies about 14/15V_{dd}. This voltage is required for the BBD chip. The current consumption of the MN3101 is 3 mA typical at 15 V. ❑

The final part of this article will be published next month.

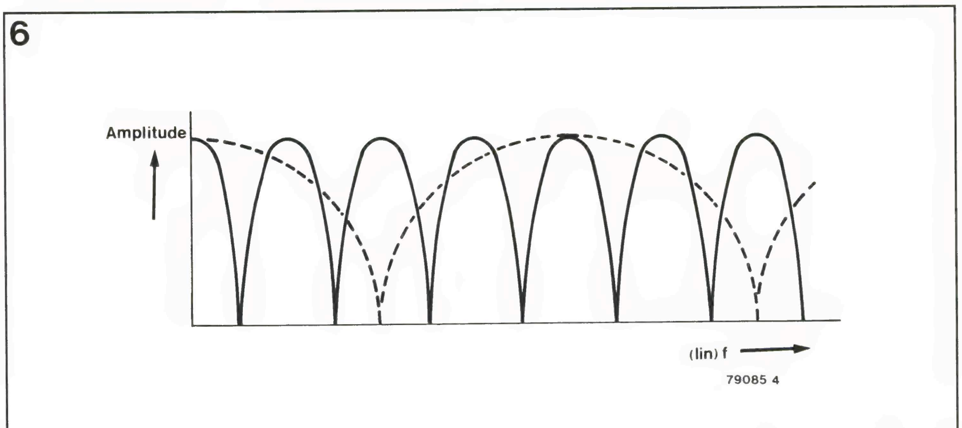

Fig. 6. Typical comb filter response of a delay line. The sound effect is 'phasing'.