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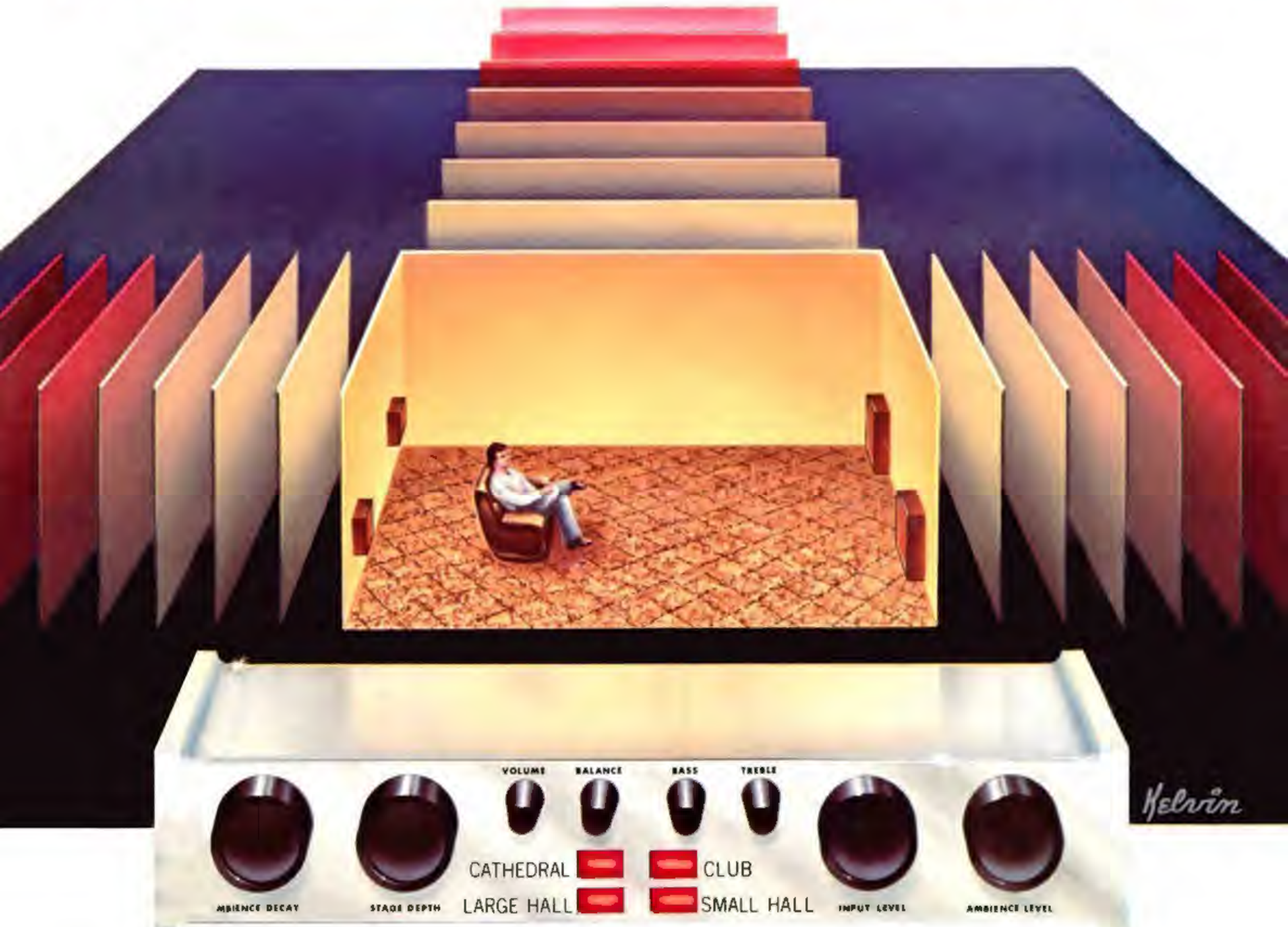

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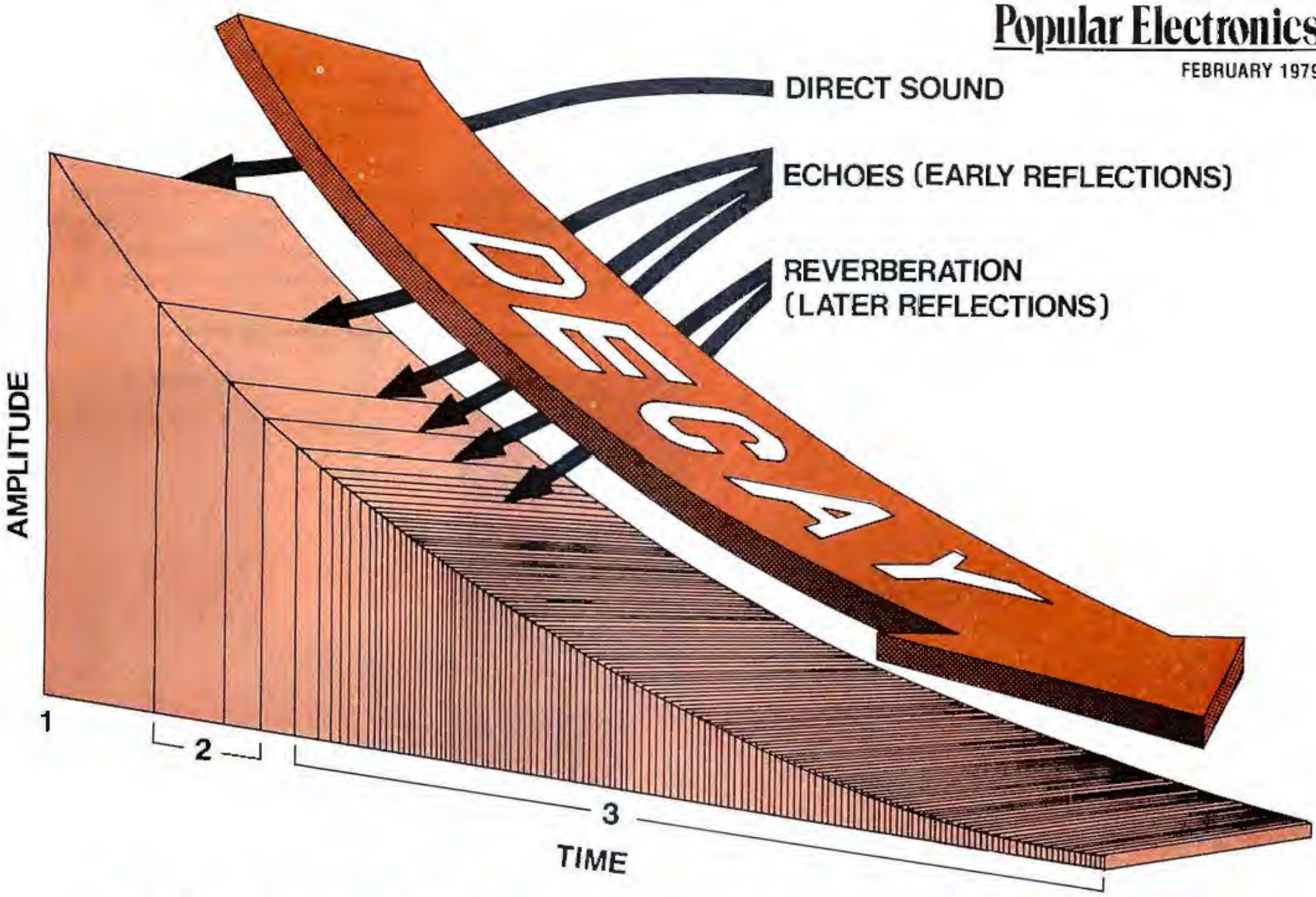


AUDIO "LISTENING-ROOM" EXPANDERS

- How time delay enhances sound reproduction
- Buyer's guide to eight models
- Performance comparisons

By JULIAN HIRSCH
Hirsch-Houck Laboratories

MOST RECORDINGS are unlikely to make you believe you are truly listening to "live" musicians. There are several reasons for this, a major one being that the ambience of a typical home listening environment is totally different from that of a concert hall or night club. In brief, indirect sounds caused by sound reflection and absorption in a large hall are different from those experienced at home. Consequently, one's listening sensation is clearly influenced by room size, room material, etc.

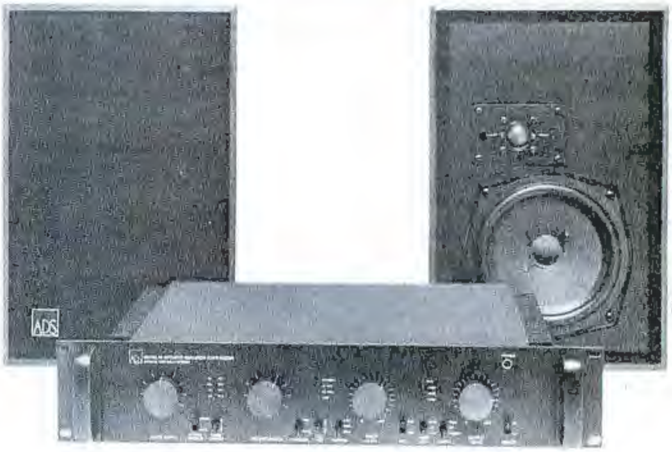
A new crop of audio components designed to apparently expand the size of a listening room insofar as hearing is concerned has aroused much interest among audio enthusiasts. All of these components introduce signal delays to simulate selectable degrees of reverberation, making reproduced program material sound more realistic. Here's a close look at each model currently available to the consumer, with an examination of special features and how well the units perform.

Reflections. The first sound to reach a listener from any point in a room arrives by a direct path. Our ability to localize the source of the sound depends heavily on this direct signal. It is soon followed by a number of reflections from the various surfaces of the room, all of which travel a longer distance than the direct sound. In general, the later arrivals have less high-frequency content than the earlier ones due to absorption by room furnishings and the audience. This combination of reflected and direct sound gives the listener a sense of the size and acoustic properties of the room, as well as his spatial relationship to the sound source. A recording that lacks these reflected or indirect sounds is dead and lifeless, but most do contain at least some of the ambience of the original environment. Ambience is often captured by suitable microphone placement or artificial means. If the latter is done properly, the result is a more pleasing and "natural" sound as compared to a recording of the same material without a satisfactory sense of ambience.

Unfortunately, the program material is usually reproduced in a room that differs drastically from the recording environment. It has its own reflection paths and delay times which are, as a rule, much shorter than those of the original location. The ambience of the listening room thus tends to *conflict* with the recorded ambience, contributing to the listener's awareness that he is hearing an artificial, "out-of-context" sound. No matter how effectively the ambience of the recording environment is captured, it will not sound natural in another room of very different size and proportions.

One of the most promising solutions

ADS Model 10



The ADS (Analog & Digital Systems) Model 10 Acoustic Dimension Synthesizer is described by its manufacturer as a "third generation" product. Built-in is a power amplifier rated to deliver 100 watts per channel to 4-ohm loads. Rounding out the ADS "package" is a pair of small speaker systems whose properties have been optimized for this application.

The Model 10 is a digital system using a proprietary form of Delta Modulation. Shift registers act as the delay storage elements. It has a choice of four short delays, selected by a STAGE DISTANCE switch. Momentarily toggling this switch up or down from its center neutral position increases or decreases the initial delay tima, respectively, as indicated by a series of LEDs on the panel. The LED string is calibrated in feet: 10, 24, 33, and 45 feet. Because sound travels approximately 1 foot per millisecond in air, these numbers also correspond to the delay in milliseconds (ms). A second group of delays, selected in a similar manner with a HALL SIZE switch, are identified as CLUB, SM. HALL, LG. HALL, and CATHEDRAL. These terms are self-explanatory. Each operation of the HALL SIZE switch simulteneously changes two delay times, the ratio of one to the other being a predetermined optimum value.

The instantaneous program level is shown by four LEDs, labelled PEAK, −10, −20, and −40, dB. An input selector switch and level control allow the user to drive the Model 10 with a wide range of available signal levels. So long as one of the dB LEDs is flashing but the PEAK LED is dark, the program level is suitable for correct operation of the system. Other controls govern the output level of the delayed signal, tha amount of reverberation introduced, and the STAGE DEPTH (a controlled injection of the delayed signal into the front channels). Toggle switches set the upper cutoff frequency of the delayad channels at 5000, 8000 or 13,000 Hz, and connect either the direct or the delayed signals to the rear speakers.

A unique feature of the ADS Modal 10 is its SOURCE AMBIENCE switch. In its MONO position, the delay circuits respond in the normal manner to both stereo and mono input signals. One problem associated with ambience synthesis is the "announcer in a cave" effect. When an amount of reverberation optimum for many kinds of music is used, an FM station announcer's voice sounds as though he was speaking at a great distance in a huge cave. This is most unnatural and disconcerting, destroying the illusion of reality that was created for the music by the time delay system.

ADS has virtually eliminated this problem by providing a mode of operation selected by the STEREO position of the SOURCE AMBIENCE switch. In this condition, the delay circuits respond only to the difference between the input channels (L-R signal). Not only does this give e very pleasing quality to much of the stereo program material transmitted on the FM band, but because announcers ere usually located "center stage", the ADS Model 10 does not delay their voices. Rather, it leaves them at front center without the disturbing effect of delay and reverberation. The system is not perfect, but it does greatly ameliorate one of the few "bugs" seemingly inherent in the concept of time delay ambience enhancement.

Separate access to the delayed outputs and the power amplifier inputs can be had via rear panel jacks. Also provided is a pair of outputs, called DELAY 2, which carry the same delay components as the regular outputs, but with a different reverberant pattern that simulates the reflections from the ceiling and rear wall of the concert hall. They can be used to drive a *second* delay power amplifier and pair of speakers, on the ceiling and rear wall of the listening room, for a further enhancement of the overall effect.

A headphone jack mounted on the front panel can drive stereo headphones with a mixture of the direct and variously delayed signal components. Plugging in the phones silences the rear speaker outputs. The power switch of the ADS Model 10 has a gradual turn-on characteristic, that brings the unit into operation over a period of several seconds in order to evoid any transient noises.

The ADS L10 speakers, supplied with the Model 10 system, are small two-way systems with 7" woofers and 1" soft dome tweeters. Each speaker is 5"H x 9.75"W x 6.5"D (38 x 24.8 x 16.5 cm) and weighs 12.5 pounds (5.7 kg).

The ADS Model 10 is completely finished in flat black and measures 15.75"W x 12"D x 3.5"H (40 x 30 x 8.9 cm). An optional set of rack-mount adapters can be installed, making the unit 10" (48.3 cm) wide. It weighs 23.5 pounds (10.7 kg). The price of the ADS Model 10 system is $995.

to this problem was quadraphonic sound. It still is, perhaps, the best solution. In theory, it is possible to capture the ambience of a recording environment in a four-channel recording and effectively reproduce it in a different listening room. The necessary time delays are "built into" the recording and allow the four speakers to recreate the acoustics of the original hall. Some quadraphonic recordings were able to achieve this goal, but recording companies soon began to concentrate on "gimmicks" and flashy effects that, however impressive, were completely unreal and unconvincing. Moreover, one had to pay for a second amplifier and another set of speaker systems. The result has been the rejection of quadraphonics by many people.

At about the same time that quadraphonics began to fade from the high-fidelity market (1976), a few small companies announced the development of time-delay ambience synthesizers. The principles on which they were based had been known for many years, but hardware was not practical until the development of suitable integrated circuits. Their purpose was to delay, electronically, the normal stereo program material (usually by several different amounts to simulate different path lengths) and recirculate the delayed signals sufficiently to approximate the effect of multiple reflections in a hall. When these delayed signals were reproduced through a second pair of speakers, located along the sides or toward the rear of the room, the "spaciousness" of a large hall could be transferred to a much smaller listening room with remarkable success.

These first-generation ambience enhancers were quite expensive, and once the amazement of experiencing their effects had worn off, their limitations became more apparent. Second- and even third-generation ambience synthesizers are now made available to the audiophile by at least seven or eight manufacturers. To varying degrees, they have overcome many of the limitations of the early units, although none is completely free of idiosincrasies. They are still expensive, and because of their inherent complexity will probably remain an expensive "add-on" to a music system. There might be some decrease in cost if delay circuits are incorporated into receivers and integrated amplifiers.

To assess the current state of the art in time delay ambience synthesis, we undertook to compare major consumer models on the market. We were well aware that this would be a very difficult task because of the peculiarly subjective nature of the entire process. Not only are measurements of an orthodox nature difficult to perform, they probably convey nothing about the strengths and weaknesses of the individual units except, perhaps, to an expert in the design of such equipment. The usual numerical test data associated with preamplifiers and power amplifiers really do not tell us much about how these devices sound. Nevertheless, we tried to do more than simply listen to them and evaluate their comparative merits.

Analog vs. Digital. Before examining any specific units, let's consider some of the basic concepts employed in designing them. The two broad classifications of time delay circuits are analog and digital. There are, of course, advantages and disadvantages associated with each. We will first look at the basics of analog delay techniques and then those that are digital.

Analog delay lines in the form of mechanical spring reverb units have been used for many years. In fact, one of the units tested (the Phase Linear Model 6000 Series Two) employs springs to obtain long delay times. State-of-the art analog delay circuits, however, employ electronic circuits in place of springs to overcome certain less-than-ideal performance characteristics usually associated with them.

Contemporary analog delays are built around integrated circuits generally known as Charge-Transfer Devices or CTDs. There are two types of CTDs, each of which is essentially a clock-controlled analog shift register. The first to be developed is commonly referred to as a "bucket brigade." This nickname comes from an analogy to the fire-fighting technique of passing buckets of water from one person to the next to move

Advent Model 500 SoundSpace Control

This is a digital PCM system using RAM storage. The delay time, selectable from 1 to 100 ms in 1-ms steps, is displayed on a large two-digit, seven-segment display whose reading of 0 to 99 is always 1 ms less than the actual delay. To adjust the delay, a spring-loaded SIZE switch is held up to increase delay, or down to decrease it. The delay increments or decrements in 1-ms steps at a rate of about 10 ms per second. Two green fluorescent bar indicators show when the program levels are within the normal operating range of the unit, and red lights flash if an overload occurs. A three-position switch selects one of three fixed input sensitivities to match the levels of the incoming signal.

In addition to the SIZE control, the only control of the Model 500 that requires regular attention is the continuous reverberation adjustment knob. Smaller knobs control volume and the bass and treble tone controls. The treble tone control injects more and more undelayed high frequencies (above 6000 Hz) into the rear outputs as it is advanced from its minimum setting. The response of the delay circuit drops off sharply above 6000 Hz. A three-position switch cuts off the delayed sound, or replaces it with the direct sound, and another switch silences the front (undelayed) program. A rear-panel DELAY switch allows the Model 500 to be used only as a delay device (no mixing or reverberation) whose two channels can be cascaded to obtain a mono delay up to 100 ms. No power switch is included because the unit is intended to be powered by a switched ac receptacle on the associated amplifier or receiver.

The Advent Model 500 SoundSpace Control is finished entirely in black, with contrasting white panel and knob markings. The cabinet and panel are rounded, making the unit look more compact than it really is. Actual dimensions are 15.75"W x 10.75"D x 3.25"H (40 x 27 x 8 cm) and weight is 10.25 pounds (4.7 kg). The price of the Model 500 is $595.

the water over a considerable distance.

In this type of analog shift register, the instantaneous amplitude of the input signal is sampled upon receipt of a clock pulse and is used to charge a small input capacitor within the IC. When the next clock pulse arrives, the packet of charge is transferred via a MOSFET to the following capacitor in the shift register, leaving the first to be subsequently charged by the next sample of the input signal. At the end of the string of capacitors, the sampled analog voltages are recovered (delayed by the time it took them to pass through many hundreds or even thousands of intermediate storage elements) and are smoothed together to reconstruct the original waveform. The total delay time is determined by the number of storage elements in the shift register and the clock frequency.

The second generation of Charge-Transfer Devices has been given the name Charge-Coupled Devices (CCD's). In many respects CCDs are similar to bucket brigades, but instead of passing packets of charge through a string of capacitors, these delay line ICs transfer *bias levels* from one MOSFET to the next. Each sample of the input signal biases a FET at a particular point on its load line, and this bias level is shifted from one FET to the next until it reaches the end of the shift register. The stream of continuously changing bias levels is converted after having been delayed into a reconstructed version of the input signal. In terms of performance, CCD's offer longer delays and higher S/N ratios as compared to bucket brigades. As before, the total delay time depends on the number of storage elements (MOSFETs rather than capacitors) and the frequency of the clock oscillator.

Digital delays use either Pulse Code Modulation (PCM) or Delta Modulation (DM). In both systems, the analog signal is first converted into digital form by an analog-to-digital (A/D) converter, whose output is a group of logic levels that define in binary form the instantaneous amplitude at the moment the waveform was sampled. In DM, the end result of the encoding is a signal that specifies the change in amplitude since the last sampling interval. This information is employed after having been suitably delayed to reconstruct the input signal's waveform.

Once the program has been digitally encoded, the binary information is transferred through a shift register composed of a series of interconnected flip-flops. The transfer is under the control of a clock oscillator; and, as with the analog systems, the total delay is a function of the number of shift register elements and the clock frequency. In some digital units, delay is achieved by storing the binary information in a random access memory (RAM) for a given length of time before being retrieved for further processing.

After the desired delay interval, the digitally encoded signal, whether stored in shift registers or a RAM, is applied to a digital-to-analog (D/A) converter. In a PCM system, the D/A converter recreates an analog version of the input signal directly. In a Delta Modulation unit, the output of the D/A converter is not a replica of the input signal itself, but rather a control signal describing the way in which the input signal has changed. This signal governs the operation of a ramp generator, whose output (after being smoothed by a low-pass filter) is the reconstructed input signal. As a practical matter, all of the commercial time delay units incorporate special precautions to keep the signal-to-noise ratio from being degraded, and to allow circuits with limited range to accommodate signals with a large dynamic range. To this end, all of them have some form of preemphasis and deemphasis, and, in many cases, a compressor/expander (compander).

Adding Ambience. As mentioned earlier, an ambience synthesizer must supply a number of different time delays in order to achieve realistic effects. Most of the models commercially available provide at least two different delays as well as provisions for recirculating them to add reverberation to the sound. This is accomplished by feeding a portion of the output of one channel back to the in-

Audio/Pulse Model One

Audio/Pulse was the first manufacturer to offer a digital time delay to the consumer and its Model One is still in production. It has been joined by the slightly less expensive Model Two, which includes a power amplifier (but no speakers).

The Audio/Pulse Model One employs Delta Modulation and shift register storage elements. Its operating controls might seem to be unconventional but they were designed to avoid the formidable appearance of a large number of knobs and switches without any loss of flexibility. The front panel contains only a row of LED level indicators and two slider controls that adjust the output levels of the delayed channels. The other controls are pushbutton switches mounted along the top front of the unit, inset slightly from the front. To set the input level within the dynamic range capabilities of the unit's A/D converters, one of six interlocking buttons simultaneously reduces the input signal and increases the gain of the stage processing the reconstructed analog signal to maintain a constant output level. A gross misadjustment of these buttons can produce either audible noise or distortion in the rear speakers, but correct adjustment is easily obtained with the aid of the LED indicators.

The Audio/Pulse Model One has four initial time delays, selected from a group of six by a pushbutton switch with SHORT and LONG positions. The available delay ranges from 8 to 94 ms. Delayed signals are recirculated for a time determined by the settings of the *five* DECAY TIME pushbuttons, which can be energized singly or in groups to produce a reverberation decay of 0.2 to 1.2 seconds. The Model One also has auxiliary outputs that can supply short- or long-delayed signals to additional speakers near the front or rear of the room to form a six- or even eight-channel ambience system.

The Audio/Pulse Model One is finished in black with walnut side panels. It measures 14.5"W x 10"D x 4.5"H (36.8 x 25.4 x 11.4 cm) and weighs 10 pounds (4.6 kg). Its price is $700.

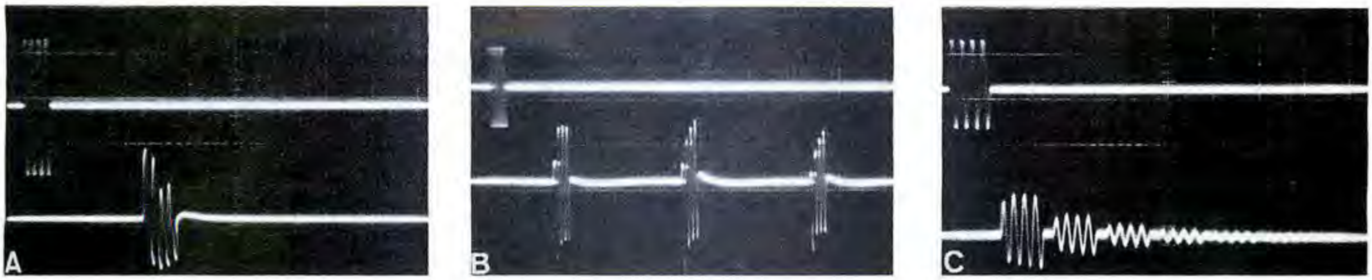


Fig. 1. Typical outputs when a single time delay is introduced (A, horizontal scale is 5 ms/cm), when multiple delays are generated (B, scale 10 ms/cm), and when a single delay is used in conjunction with recirculation to simulate decaying reveration (C, scale 5 ms/cm).

put of the other channel. The result is a decaying reverberation that can take a second or more to fade away, as it would in a real concert hall. Scope traces of typical time delay outputs with one delay, multiple delays, and a single delay interval with recirculation are shown in Figs. 1A, 1B, and 1C respectively. The amount of reverberation that should be added is a function of the original signal, as well as personal taste. A "dry" recording will benefit from considerable reverberation while a "live" sounding recording will be muddied and confused by adding too much.

One fundamental difference between the units we tested is the coherence of their delayed outputs (or lack of it). When a mono signal is fed to both inputs of a delay unit, and the two delayed outputs are in phase, they are said to be *coherent*. Some designers feel that this is a desirable quality, while others hold that there should be no fixed phase relationship between the two delayed outputs. This is based on the assumption that the multiple reflections in a real concert hall do not have any defined phase relationship. Consequently, some of the delay units contain "randomizing" circuits that give completely random phase between the two outputs. Oscilloscope traces of coherent, partially coherent, and noncoherent outputs are shown in Fig. 2.

Connecting an ambience enhancer to a stereo system is a fairly straightforward process, and all are treated in the same manner. The best place to connect them is between the preamplifier output and the power amplifier input so that the preamplifier's volume control will affect all output signals. This point is accessible in many receivers and integrated amplifiers. The time delay unit provides both front- and rear-channel outputs. The front-channel signals are returned to the power amplifier inputs of the integrated amplifier or receiver. In most models they pass through a straight-wire connection, though in some cases there is an active stage in which some of the delayed signals can be mixed with the front-channel signals.

The rear (delayed) channels require a separate stereo power amplifier, but not one with elaborate control facilities. Its power output, as a rule of thumb, should be roughly one half that of the "front" amplifier. In a few of the units we tested, the power amplifier is built in. However, those audiophiles who have a spare amplifier on hand will not accord this feature as much weight as those who don't. The delayed channels also call for an additional pair of speakers, whose optimum placement and performance characteristics are a matter of debate among users of time delays.

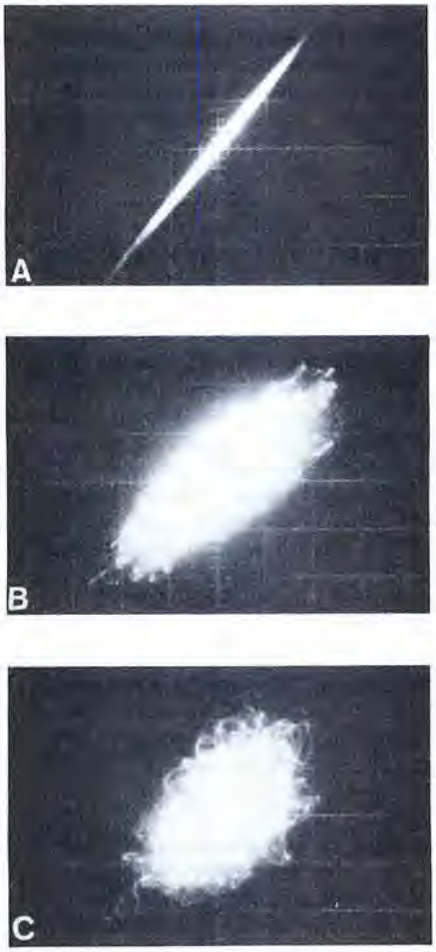


Fig. 2. Some time delay components have coherent (in-phase) outputs (A) as in Bozak Model 902. Others, such as the Advent Model 500, are partially coherent (B). Still others, such as the Audio/Pulse Model One, are noncoherent (C).

It is most generally held that the rear speakers should be along the side of the room, preferably in front of the listener. Wherever they are placed, it is most important that they are not heard as distinct sound sources (sometimes placing them at a wall/ceiling junction, or on the floor facing upward, will produce the most suitable results). To further ensure that the rear speakers cannot be aurally localized, wide dispersion is desirable. Extended frequency response is not needed, because most of the time delay units have restricted high-frequency response. This corresponds to the reduced high-frequency content of natural reverberant signals. One of the most interesting subjective effects of time delay ambience synthesis is its apparent enhancement of the audio system's bass response. It also makes the overall program sound louder, without in itself having much real bass or contributing much to the acoustic output of the system.

At least two of the manufacturers who build power amplifiers into their time delay units also supply speakers with the system, but the units can be purchased without speakers if desired. In general, the systems we tested are priced very comparably when allowance is made for the cost of an amplifier and pair of speakers. Although some of the accessory delay units have a full complement of controls, the system will, after initial set-up, normally be controlled entirely by the main stereo preamplifier or integrated amplifier. Once the controls of the time delay unit have been set for the desired effect, they will need re-adjustment only to accommodate different types of program material. One of the inherent disadvantages of time delay ambience synthesis is that the optimum amount of delay and reverberation is not the same for all types of program material, so a certain amount of fussing with the controls will be necessary when shifting from one type of material to another.

Having examined the basic operating principles of both analog and digital time delay units, the reader should take a close look at each of the eight products available to our test lab, as they are described in the accompanying boxes.

Test Procedures. The data supplied by the various manufacturers, either in the form of specifications or actual measurements, ranged from virtually nothing (SAE) to extremely complete test information (from ADS). Rather than attempt to duplicate their figures, we preferred to accept any manufacturer's rating as valid (especially since almost all of the information would be difficult for anyone not highly skilled in the techniques involved to understand or interpret).

We limited our measurements to frequency response under various delay and other operating conditions, and in most cases, the output noise and distortion. Driving one input with a 4-cycle burst of 1,000-Hz sine waves, we examined the delayed and reverberated outputs on an oscilloscope, to see how densely the delayed and reverberated pulses were "packed" or if the burst shape was altered materially. One of the more informative tests was to drive both inputs from a pink noise source, and connect the two delayed outputs to the X and Y axes of the scope. This shows whether the delayed outputs are coherent, partially coherent, or completely phase randomized.

Perhaps the most meaningful test one can make on an ambience synthesizer is to listen to it, to hear what it does *to* and *for* the sound, and to learn how easy it is to adjust for most pleasing results. Although this is completely subjective, it is still possible to compare different units in listening tests. The ADS Model 10 comes with its own amplifier and speakers, and was set up in a different room where it could not be compared directly to the others. The Bozak Model 902 was made available to us for only a few hours, so our evaluation was very brief. However, we bypassed its power amplifiers and did not use its speakers so that its delay circuits could be evaluated on a more or less even basis with the other models.

For a listening comparison, we set up all the time delay units with their inputs in parallel, driving them with the same preamplifier. Their delayed outputs were connected to high-level inputs of an integrated amplifier that drove our rear speakers (a pair of AR-7's, mounted at the wall-ceiling boundary). The front speakers were usually AR-LST's, although some other types were also used. We tried to adjust the time delay units for roughly similar operating conditions, but this was not really possible because they each have different combinations of delayed signals and many are not calibrated. We did match their rear

Audio/Pulse Model Two

This unit was received too late for complete testing, but was set up for comparative listening tests with the other units.

A second-generation time delay, the Model Two employs some of the circuitry of the Audio/Pulse Model One, especially its "delta modulation with memory" used to convert the analog input signal to digital form. However, storage of the digital information is accomplished with a RAM rather than shift registers as in the Model One. Furthermore, the Model Two has built-in power amplifiers rated to deliver 25 watts per channel into 8-ohm loads from 40 to 8000 Hz with no more than 0.5% THD. The power bandwidth of this amplifier is much more restricted than that of even a modest contemporary component amplifier. It must be remembered, however, that the upper cutoff frequency of the delayed channels is *only 8000 Hz* so the limited bandwidth of the amplifier section does not compromise the performance of the unit as a whole.

In appearance, the Audio/Pulse Model Two is more like other second-generation delay units and bears little external resemblance to the Model One. It is about the same size as the Model One (but an inch lower in height), finished in flat black, with rounded corners on the cabinet. Its general styling is not unlike that of the Advent Model 500. In place of the pushbutton switches used in the Model One, the Model Two has knob controls for most of its functions. A row of LEDs to monitor input level is used in conjunction with an INPUT LEVEL potentiometer. This control is not ganged with an output level control (as is accomplished by the pushbutton system of the Model One), so it is necessary to readjust the OUTPUT LEVEL control when the input level is changed. Bass and treble tone controls affect only the delayed signals. Once adjusted to suit the requirements of the delayed channel speakers and the room layout, they need not be changed. The operation of the unit is governed by a main FUNCTION control, a four-position switch marked SHORT DELAY, DEFEAT, LONG DELAY, and DIRECT. In the SHORT DELAY mode, the signal undergoes initial delays of 19, 33, and 51 milliseconds. The LONG DELAY mode gives delays of 39, 66, and 103 milliseconds. In DEFEAT, the delayed outputs are silenced, and in DIRECT, they are driven with an undelayed signal. Like the Audio/Pulse Model One, the Model Two has incoherent outputs, bearing a random phase relationship to each other.

The remaining major control is a horizontal slider marked AMBIENCE. This adjusts the mixing weights of the fully and partially delayed signals as they are recirculated through the system. At MIN, the shorter, partially delayed signals are emphasized, but at MAX the fully delayed ones are emphasized and maximum reverberation is added. The rear apron of the unit accommodates circuit breakers which protect the rear-channel speakers. The INPUT phono jacks accept a signal from the preamplifier output or a tape monitor loop. Two jacks carry an unmodified front channel signal back to the main amplifier and a DIN socket is provided for driving the Audio/Pulse Two from the front speakers if this is more convenient. Finally, delayed outputs are furnished for driving an external power amplifier if more power than is available from the built-in amplifier is required.

The Audio/Pulse Two is 15"W x 10½"D x 3½"H (38.1 x 26.7 x 8.9 cm). It weighs 14 lb (6.4 kg). The price is $539.

channel levels, however, by switching off the front speakers and listening to the delayed outputs as we switched from one unit to another. Also, a cassette recording was made of the delayed output of each unit; using the same program material under various adjustment conditions, so that we could hear the delayed sounds of all the units under controlled conditions.

For the most part, however, we simply listened to records and FM broadcasts, selecting one or another delay unit to see how they could be adjusted to enhance the sound (or how they could degrade it if incorrectly adjusted). This procedure took several weeks, and left us with some fairly solid conclusions and many more unanswered questions.

Test Results. The test data (see Table) shows that, for the most part, the frequency response ratings of the delay circuits of the tested components are quite accurate. Signal-to-noise data is less easily correlated with published ratings because most are based on A weighting, which reduces the output noise voltage below our 100-microvolt minimum. (−80 dB re 1 volt). Also, the noise of most units varies somewhat with different control settings. The important thing to know about their S/N performance is that noise cannot be heard in the output of any of the tested units when operated in accordance with the manufacturer's instructions. It is not even possible to operate most of them incorrectly in this respect. (The Audio/Pulse Model One *can* be set up so that hiss can be heard, but it requires a deliberate effort.)

The measured THD in the delayed outputs varies over a wide range, from a small fraction of a percent to about 2 percent at a 1-volt output level. This is of no practical significance because the sound from the rear speakers is never heard as a separate source if the system is adjusted correctly. Furthermore, the distortion is composed entirely of low-order components and thus is not audibly offensive even when one listens only to the delayed channels.

Two genuine points of distinction can be seen among the tested units, however. The upper frequency limit of the delayed sound varies from a few thousand to 13,000 Hz. In the analog delay lines, the highs roll off appreciably as delay is increased, although this can be compensated for with suitable circuits as in the Sound Concepts SD550. The digital systems have a constant bandwidth *independent* of the delay setting. They do not necessarily have *wide* bandwidths, merely constant ones. The Advent Model 500, for example, has a fixed 6000-Hz bandwidth, and the Audio/Pulse Model One has a fixed 7000-Hz bandwidth. The ADS Model 10 has an impressive 13,000-Hz bandwidth at its maximum setting. ADS does not suggest using the full bandwidth of its unit on any but the finest program material, and in practice there seems to be little reason to go above 8000 Hz or so in light of the spectral content of the naturally occuring reverberant signals sought to be recreated.

The coherence of the delayed channels varies from being fully in-phase as is the case with the Sound Concepts, SAE, and Bozak units through partially random in the Advent to fully random in the ADS, Audio/Pulse, and Phase Linear models. Each of these companies claims that *their* approach is the most correct one, so it is difficult to be dogmatic about the matter. Logically, it would seem that multiple reflections will never be in phase on both sides of the room, suggesting that a coherent-output device is inherently incapable of simulating real concert-hall ambience. Offsetting this is the fact that the program material on one stereo channel is not in phase with respect to that on the other (if it were, the program material would be monaural), so that even a "coherent"

Bozak Model 902

The Bozak Model 902, which includes a 35-watt/channel power amplifier, is a second-generation product from that company. Its delay is derived from the latest type of analog shift register, a charge transfer device capable of longer delays and much better signal-to-noise characteristics than the original CCD (charge-coupled device) bucket brigades. The delay time is continuously variable by a large front-panel knob from about 20 to 120 ms. Another control of the same size adjusts the output level. The other controls are slightly unconventional but highly effective.

A small SIGNAL BLEND control provides only delayed signals to the speakers at its extreme clockwise setting, and only undelayed (direct) signals at its fully counterclockwise position. Between these extremes, it varies the mix of direct and delayed sound to the rear speakers in "pan pot" fashion. Reverberation is governed by a DELAY REMIX control which gives a smooth transition from a rather "dead" sounding room to one that is very "live". The TREBLE CONTOUR control boosts or cuts the high-frequency response of the delayed channels to suit one's taste. At a 25-ms delay, the response of the delayed channels is flat to about 6000 Hz, decreasing to approximately 2000 Hz at 120 ms.

A novel level monitor at the right of the panel spans about 40 dB in two 20-dB ranges. It is an auto-ranging LED bargraph display whose scales shift automatically with changes in input level. The level in each channel is seen as an expanding and contracting vertical red bar whose height is calibrated by an adjacent scale. A 0-dB reading is the maximum recommended input, and the word CLIP appears on the display if that value is exceeded.

The Model 902 has a built-in power amplifier rated at 35 watts per channel into 8-ohm loads. It is supplied with a pair of Bozak Model DS1800 speakers, which are compact floor-standing units having upward-radiating, essentially omnidirectional outputs. Low-level delayed outputs are also available at panel jacks for use with another amplifier. The delay circuits are also packaged without the power amplifier and speakers as the Model 901.

The Bozak Model 902 is finished in black, with its basic performance curves and a functional block diagram screened in white on its top cover. It measures 17¾"W x 11¾"D x 2½"H (45 x 29.8 x 6.4 cm) and weighs 14¼ lb (6.5 kg). The price of the complete system, including speakers, is $975. The Model 901 Delay Unit alone (less amplifier and speakers) is $625.

delay unit would have random phase between its delayed outputs. Probably, the distinction between these types of systems will be apparent only in the way they treat mono programs, and our listening tended to confirm this.

One *cannot* help wondering about the amplifier power requirements of the delayed channels. Bozak's amplifier is a modest 35-watt/channel (into 8 ohms) unit, but ADS can deliver 100 watts per channel into 4-ohm rear speakers. Does this seem reasonable in view of the fact that the rear speakers are never to be audible as sound sources? Yes, because they are usually played at only slightly lower (perhaps −3 to −6 dB) levels than the front speakers. They are not heard directly because of the Haas effect, which causes us to localize the sound from its first arrival at our ears. This leaves no doubt where the sound is coming from (the front speakers), and the sound from the delay speakers arriving a number of milliseconds later merely adds ambience and is not sensed as the same program. (If the direct sound is fed to the rear speakers, the change is dramatic, yet the power level of the rear channels might not change at all.)

Conclusions. We come now to the specifics of our evaluation of time delay ambience enhancers. Ultimately, opting to add ambience enhancement and the

Phase Linear 6000 Series Two

This unit employs analog bucket-brigade devices as well as mechanical (spring) delay lines for long reverberation times of up to 4 seconds. Most of its control operations are handled by eighteen pushbutton switches grouped functionally. Large knobs control the front and rear speaker volume (separately), the application of either direct (undelayed) or delayed signals to the rear speakers, and select one of four rear-channel, frequency-response characteristics. In addition to a nominally flat response, the switch provides LO or HI CUT response or both simultaneously.

The Phase Linear unit has two primary delay times, one 15 or 20 ms and the other 60 or 90 ms, depending on the clock frequency. Each is selected individually, with a choice of three signal levels (+3, 0, or −3 dB). There ara two groups of recirculated signals, identified as SHORT and LONG (signifying that they have been recirculated through the 15- or 20-ms or the 60- or 90-ms primary delay register, respectively). The levels of the recirculated signals can also be set at +3, 0, or −3 dB.

The REVERBERATION controls affect the signals that are passed through the mechanical delay line. The SOURCE buttons derive the inputs to the reverberation section from either the SHORT or LONG primary delay registers, and the MASTER CLOCK pushbutton changes the clock frequency by about 20%. This affects all time delay functions in the Model 6000 and is thus a fine adjustment for the entire delay circuit. Taps on the mechanical delay line are selected by the TIME pushbuttons, giving reverberation times of 1, 2, or 4 seconds. There are no overload or other level indicators on the Model 6000, but its dynamic range is sufficient to accommodate the normal output of a preamplifier.

The Phase Linear Model 6000 is styled to match other Series Two Phase Linear components, with a pale gold panel and matching knobs. It is 19"W x 10"D x 5.5"H (49.3 x 25.4 x 14 cm) and weighs less than 20 pounds (9.1 kg). The price of the Model 6000 is $600.

Measured Performance

Model	HF response @ −3 dB re 1 kHz (kHz)	S/N unwtd re 1 volt output (dB)	THD @ 1 V. output (%) 1 kHz	Input volts operating range	Coherent delay outputs	Comments
ADS	12.2	66	0.14	NA	No	Built-in Amplifier (100 W/channel into 4 ohms)
Advent	6 (undelayed can be boosted to +11 dB in rear outputs)	72	0.3	0.015-3.0	Partial	
Audio/Pulse Model One	7 (CONTOUR can boost 10 dB in 40-60 Hz)	54-63	NA	NA	No	
Audio/Pulse Model Two	NA	NA	NA	NA	NA	
Bozak	7.7	60 (more than 80 with A-wt)	0.5 @ 0.25 volt out (1 v. in)	0.03-1.75	Yes	Built-in amplifier (35 W/channel into 8 ohms)
Phase Linear	4.5 SHORT	More than 80	2 @ 1v 0.45 @ 0.3v	3 max.	No	
SAE	5	57-60	0.2 @ 1.8v. Much higher at LF	0.08 or more	Yes	High regeneration puts large 400-Hz component in output
Sound Concepts	8	70	0.8	NA	Yes	

choice of a particular model are subjective decisions. Our experience with these devices has convinced us that conventional measurements and specifications are of little or no value to an audiophile making a choice from among these products. If the usual undesirable contributions of distortion and noise cannot be heard (and they cannot, by any stretch of the imagination, in any of the units we tested), then all that is left that should influence the choice is their combinations of available time delays and reverberation, and to some extent their delay-channel frequency response properties.

We suspect that a multiplicity of delay times might give a more realistic simulation of concert hall sound than a simple single delay plus reverberation. This is probably true, yet nothing in our experience convinces us of that fact. Plainly put, every one of the tested units is capable of providing a *tremendous* enhancement of the natural qualities of sound. A good time delay system can do more to improve the realism of home music reproduction than any other $1000 investment we can imagine (assuming that one already has good front-channel components).

The other side of the coin is that virtually every one of the units tested can also create a terribly unnatural, obviously artificial sound when misadjusted. Some are more capable of such misadjustment than others. This should not be held against them because they can just as easily be adjusted correctly. For example, the Phase Linear Model 6000 can be set to give bizarrely long

SAE Model 4100

This component is so new that specifications and a complete instruction manual were not available when we received the unit for evaluation. The preliminary instruction manual, apart from a basic description of the unit's controls, told us only that the Model 4100 has three separate delay times, labelled SHORT, MEDIUM, and LONG roughly corresponding to delays of 10, 30 and 50 ms.

Except for three pushbutton switches, all controls are horizontal slider potentiometers. A DIRECT pushbutton feeds the input signal, undelayed, to the rear outputs when it is engaged. When the switch is disengaged, the rear outputs furnish the delayed signals. The Model 4100 is designed to be compatible with quadraphonic systems and has input jacks for externally derived rear-channel signals.

Pressing the DISCRETE pushbutton routes the externally derived rear input signals directly to the output jacks, allowing full quadraphonic operation if the unit is connected to a four-channel system. Finally, the BLEND pushbutton injects some delayed signal into the front channel outputs to improve ambience when listening to some types of closely miked program material. For normal ambience enhancement, all three pushbuttons are left in their OUT positions.

An INPUT LEVEL slider matches the unit's sensitivity to the incoming program material with the aid of a red PEAK LED. Once this control has been adjusted so that the LED does not flash on the loudest program peaks, there is usually no need for further adjustment. Each of three delay circuits has its own level control slider, and the outputs of the three are added according to the settings of the SHORT, MEDIUM, and LONG level controls. The REGENERATION slider governs the amount of delayed signal that is recirculated through the delays to produce a reverberative effect. The final control is an OUTPUT LEVEL adjustment. Because the three delayed signals are summed, any substantial change in the summer control settings may require readjustment of the output level control which affects only the delayed channels.

The SAE Model 4100 has distinctive SAE styling with a black cabinet and walnut side panels. It measures 15.75"W x 8.4"D x 3"H (40 x 21.3 x 7.6 cm) and weighs approximately 7 pounds (3.2 kg). The price of the Model 4100 is $500.

Manufacturers' Specifications

Model	No. of initial delays	Range of initial delays (ms)	Longest delay (ms)	Reverb. time (s)	Input sens. range (volts)	S/N (dB)	Delay bandwidth (kHz)
ADS	3	10-40	100	0-1.6	0.75-3	80	13
Advent	2	1-100	100	NA	0.3-3	80	6
Audio/Pulse Model One	4	8-94	94	0.2-1.2	0.14-2	65	8
Audio/Pulse Model Two	3	19-103	103	0.1-0.6	0.05-3.3 V (low) 1.2-60 V (high)	72	8
Bozak	1	20-120	120	up to several	NA	NA	7.7
Phase Linear	2	15-90	90	0.2-4	2.5 max	94	6 (short) 2.5 (long)
SAE	3	10-50	50	NA	NA	NA	NA
Sound Concepts	1	5-50 (100 in mono)	50 (100 in mono)	NA	NA	85	8

Sound Concepts Model SD550.

One of the early time delay devices for home music systems was a Sound Concepts product, and the current Model SD550 is a second-generation device. A bucket-brigade delay system, it is designed for use with four-channel as well as stereo systems. Its operation is controlled by five sliders and a group of four rocker switches. A DELAY TIME control has a calibrated range of 5 to 50 ms. The REVERBERATION slider calibrated over an arbitrary range of 0 to 10, governs the amount of signal injection from the delayed output of one channel to the input of the other.

The high-frequency response of any bucket-brigade system drops off rapidly as delay time increases. Fortunately, this effect is consistent with the natural increase in absorption of highs as one moves back in a concert hall and longer delays are involved. It therefore does not result in any unnatural or grotesque effects. Nevertheless, one school of thought holds that wide bandwidth is desirable or even necessary in the delayed channels, and Sound Concepts has provided a means to achieve this result. The HI FREQ ROLLOFF slider is calibrated in decibels from +6 to −3 dB, and has a separate scale of 5 to 50 matching the delay time scale. When it is set to correspond to the delay in use, the high frequencies are boosted to maintain the frequency response out to approximately 8000 Hz, where it is down 3 dB nominally.

A FRONT MIX LEVEL slider controls the amount of delayed signal that is injected into the front outputs of the SD550 and a REAR LEVEL control governs the overall output of the delay channels.

The DELAY RANGE rocker switch parallels the two input channels and cascades the delay channels to provide a mono output delayed up to 100 ms. The REAR OUTPUT switch routes either the delayed channels or the back channels of a gradraphonic system to the rear outputs. A DELAY MIX rocker switch allows the delayed outputs to be injected into the front channel outputs to a degree determined by the setting of the FRONT MIX slider control. The INPUT switch enables the delay processing to be applied to the usual stereo or mono input signals or to the front channels of a quadraphonic system. Sound Concepts suggests that the latter alternative will often given a more natural result than that obtained using a standard four-channel decoder.

The Sound Concepts Model SD550 is exceptionally noncritical in its level requirements. Included is a 2:1 compander that reduces internal noise to negligible levels. There is no risk of overloading it from any normal preamplifier output signal. In fact, the unit has no obvious level controls, unlike most other ambience enhancers. Actually, there are two screwdriver-adjustable input level controls in the rear, together with overload LEDs, but they are preset and it is best not to disturb them.

The delay unit is finished in flat black, measures, 15.5"W x 9"D x 3.5"H (39.4 x 22.9 x 8.9 cm) and is also available with a rack mounting panel 19" (48.3 cm) wide. It weighs 7 lb (3.2 kg) and is $675.

reverberations which are intolerable with most kinds of music. With some choral works, however, the effect is uncannily like "being there". At the other extreme is the ADS Model 10, which is the most subtle of the group in the nuances of its sound modifications. It is difficult to make anything sound really wrong through this unit, the only one to have solved, at least partially, the "announcer in a barrel" sound effect.

If all these units work so well, and sound so much alike, are there any real distinctions to be made between them? Yes, indeed, but first let it be said that they do *not* all sound alike. True, most of them sound more alike than different, but the one that could never be mistaken for any other is the Phase Linear. Its drastic loss of highs and general emphasis on mid-bass output make it sound extremely muffled and heavy when heard without any contribution from the direct channel. Nevertheless, when used as a time-delay device, its sound is not as objectionable.

The *real* distinctions are in the area of human engineering. That is, how easy or difficult the units are to set up and adjust for the desired effect. Here we have some definite conclusions, although they must be qualified as being purely subjective.

The easiest and most logical time delay device to adjust and use, without question, is the Advent Model 500 SoundSpace Control. The single-lever SIZE control, and the small REVERBERATION knob, are the only controls that must be touched on this unit, once the initial installation has been made. The unit is so noncritical in matters of level adjustment that we consider it to be close to a "set and forget" device. In spite of this, it has all the flexibility most would ever want, and sounds superb.

The most refined and sophisticated of the group (referring to its capabilities, not necessarily its internal design) is the ADS Model 10. It has, in our view, an overabundance of controls, most of which have such subtle effects that we frequently could not detect them. They do give the panel a cluttered appearance, in spite of the use of miniature toggle switches for many of them, and we never could manage to handle the unit without accidentally disturbing the setting of something. This complexity is offset to a great extent by the superior sound from the Model 10. It was especially impressive on mono discs, which acquired an ambiance that put to shame most current stereo records when they were played through the time delay system. Also, the speakers of the Model 10 are small, unobtrusive, and deliver an excellent sound quality.

Although we had a very limited exposure to the Bozak Model 902, we could see that it is one of the simplest to use, and probably comes close to the Advent 500 in that respect.

The Audio/Pulse Model One has been a part of one of our music systems for some time. Its performance as a time-delay device is first rate, but the need to reset the input level switches every time a large change in signal level occurs is disconcerting. Also, operating the decay-time and level pushbuttons can sometimes introduce an audible "twang" in the sound.

With respect to sound quality and delay effects, the Audio Pulse Model Two appears to be very comparable to the Model One and several other delay units. That is, it can give a very satisfying sense of ambience or can be set for an exaggerated and unnatural echo sound or any condition in between.

The major difference from the Model One, from the user's standpoint, is the vastly simplified operating control configuration of the Model Two. It is effectively almost identical with the Advent unit in its control complexity and ease of use, except that the level indicator LEDs of the Audio Pulse give the impression of requiring more frequent level adjustments as program conditions change. Because the output level must be adjusted at the same time, this is rather awkward compared to the three-position input sensitivity switch of the Advent, which hardly ever requires attention. However, we found the setting of the sensitivity to be much less critical than that in the Model One; and so long as some of the green LEDs are flashing, noise in the output is inaudible.

Considering the substantially lower price of the Model Two compared to the Model One, and its greatly simplified operation and built-in amplifier, it is clearly an outstanding value. To our ears, it was not quite as undetectable in operation as the Advent or ADS units, but the difference was slight.

The Sound Concepts SD 550 has also been in use here for some time, and we find it to be one of the easiest to set up. Not only does it not make any noise, but we have yet to be able to flash its rear-mounted overload lights. Despite its relative simplicity as to the number of available delays and their processing, it sounds natural so long as maximum reverberation is not used. This can introduce a "boing" sound sometimes.

The SAE Model 4100 is especially interesting because of its price which is appreciably less than the others. It has all the versatility one would desire and sounded, for the most part, as good as any of the other units in our comparative tests. On the human-engineering side, we suspect that a proper instruction manual would overcome most of our objections. We set the controls by guesswork, and could not fault the results. Our two criticisms deal with first the manner in which the three delay signals are summed, which affects the overall volume of the delayed sound when the delays are changed, and second the regeneration control, which can produce a hard, "twangy" sound when set too high. If its setting was limited to the lower half of its range, it sounded fine. We note also that, although it is a bucket-brigade unit, the delays are fixed and apparently compensated for a uniform frequency response, thus, the upper frequency limit remains about 5000 Hz no matter where the level adjustments are set.

Finally, we come to the Phase Linear Model 6000. We are "turned off" by the prospect of punching our operating conditions into a panel of eighteen pushbuttons. In fairness, it is not as difficult to use as it seems, once one has had some practice, but it is still the least convenient of the group to operate. We also find its lack of highs sometimes noticeable and objectionable. Although the unit has a knob that can reduce the highs, we cannot imagine the need for this. Rather, it could use a considerable treble boost. Perhaps the ease with which the Phase Linear can be set to give unnatural effects was another factor that influenced our reaction to it.

Whatever type of time delay device is used, the most important thing to remember is "IF YOU CAN HEAR THE REAR SPEAKERS, THEY ARE TOO LOUD!!!" The delayed volume can be turned up until it is audible, and then backed off until the rear channels can no longer be heard as distinct sound sources. Check by shutting off the rear speakers. (Many of the delay units have a switch for this. It can be done at the amplifier.) If the level is correct, shutting them off will cause the sound to contract to the front, becoming dull and comparatively lifeless. Restoring the ambience will make you wonder how your ever got along without it! ◇

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Bozak, Inc.
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Phase Linear Corporation
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Scientific Audio Electronics, Inc.
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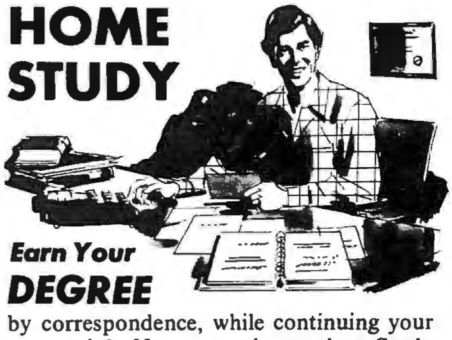

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