Moylan

EFFECTRON II

OWNER'S MANUAL
ADM 64

DeltaLab

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INPUT

* INPUT LEVEL The input level control adjusts the signal level and consequently the thru put gain of the system

At the MIN setting the system gain is (x 0) to allow high level signals (up to 7.1 Vrms) to be processed.

At the MAX setting the system gain is +26dB (x 20) to allow low level signals (down to 0.1 Vrms) to be processed. The output of the unit will be amplified by a factor of 10 with respect to the input.

When the control is at the 12:00 o'clock (straight up) position, the system gain is approximately 6dB (x 2); i e., the output signal will be the same amplitude as the input signal.

* LIMIT: The Red LIMIT LED indicates that the signal level being processed is at the OdB reference point. OdB indicates the onset of clipping and/or slew limiting There is an additional 6dB of headroom beyond what is indicated to allow for unanticipated overloading in actual usage.

* ACTIVE: The green ACTIVE LED indicates that the signal being processed is greater than -20dB below reference. The INPUT LEVEL should be set such that the green ACTIVE LED is full on for most of the time. If the input is too low the green ACTIVE LED will be off for most of the time.

FEEDBACK

The FEEDBACK control varies the amplitude of the signal that is fed back and regenerated. Close to 100% feedback, just short of oscillation, is possible at either of the extreme settings.

(-) At this setting, the signal is fed back out of phase (negative feedback) The maximum setting is full counter clockwise; various amounts of negative feedback are possible starting from zero through the maximum position

(+) At this setting, the signal is fed back in phase (positive feedback) The maximum setting is full clockwise, various amounts of positive feedback are possible starting from zero through the maximum position

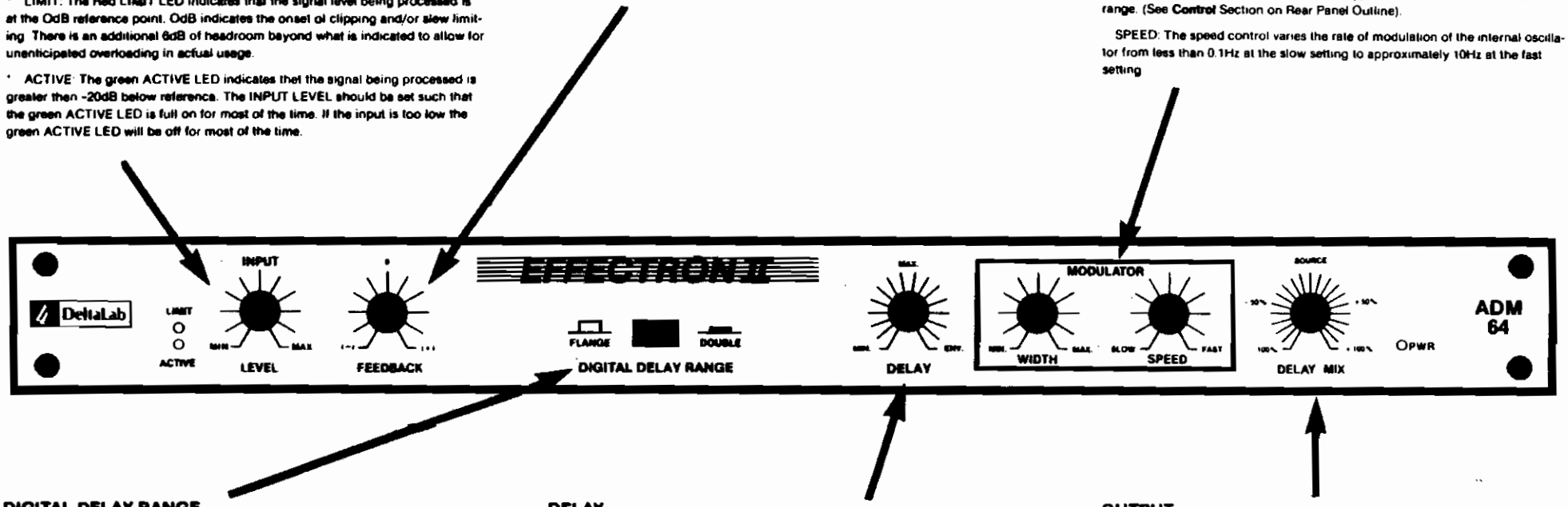
0 At the zero (click) center position, there is no feedback — hence, no regeneration

MODULATOR

The ADM64 has a precision internal VCO (Voltage Controlled Oscillator) which is controlled by an LFO (Low Frequency Oscillator). The LFO controls make up the modulator section of the front panel. These controls interact with the DELAY front panel control and with the CONTROL input located on the rear panel.

WIDTH The width control varies the sweep of the internal oscillator In the max position, the delay times vary over an 8 to 1 range, allowing three octaves of sweep. Should any setting less than MAX be used, the DELAY control should be adjusted to set the desired center of the sweep The WIDTH control should be set to the MIN position if the LFO is not being used The WIDTH control also interacts with any externally applied control signal and should therefore be set at MIN for maximum control from an external source, such as a footpedal, synthesizer, external envelope follower or other similar type control in the -0.62 to +6.2v range. (See **Control** Section on Rear Panel Outline).

SPEED: The speed control varies the rate of modulation of the internal oscillator from less than 0.1Hz at the slow setting to approximately 10Hz at the fast setting



DIGITAL DELAY RANGE

FLANGE: The pushbutton is "out" to select the time delay used for flanging effects. Using the modulator controls, the range varies as follows:

.9ms to 7.2ms at the MAX setting of the WIDTH (8 to 1) control.

And, using the delay control, the range varies as follows:

1.0ms to 4.0ms from MIN to MAX of DELAY control (4 to 1).

When the delay is between MAX and ENV, the internal envelope follower modulates the flange over a maximum range of 4.0ms to 1.0ms at the extreme setting with wide dynamic range input signals. Of course, by using combination settings of both the delay control and modulator controls, various flange effects are possible.

DOUBLE: The pushbutton is "in" to select the desired range suitable for doubling effects. The delay range varies this from 16ms to 64ms. The delay control can be used to "detune" the doubling in either the MIN-MAX range of the MAX-ENV range in conjunction with the modulator controls. Be aware that there is a mixing of modulator signal into the intermediate setting of the delay control

DELAY

The delay control adjusts from minimum to maximum (4 to 1) over the counter clockwise half of the control The delay control also activates an Envelope Follower signal in the clockwise half of the control. This signal is used effectively in both the flange and doubling modes

MIN: In the flange mode the delay is 1ms.
In the double mode the delay is 16ms.

MAX: In the flange mode the delay is 4ms.
In the double mode the delay is 64ms.

ENV: In this mode the delay automatically follows the amplitude envelope of the audio being processed

Intermediate settings are possible

OUTPUT

The output level is not adjustable The output signal level varies only with the input signal level and input level control

DELAY MIX. The processed signal can be mixed in or out of phase with the source via the delay mix control.

. At -100%, the output consists of only the delayed signal (out-of-phase)

At -50%, the output consists of equal amounts of source and out-of-phase delayed signal

. At SOURCE, the output consists only of the source signal

At +50%, the output consists of equal amounts of source and in-phase delayed signal

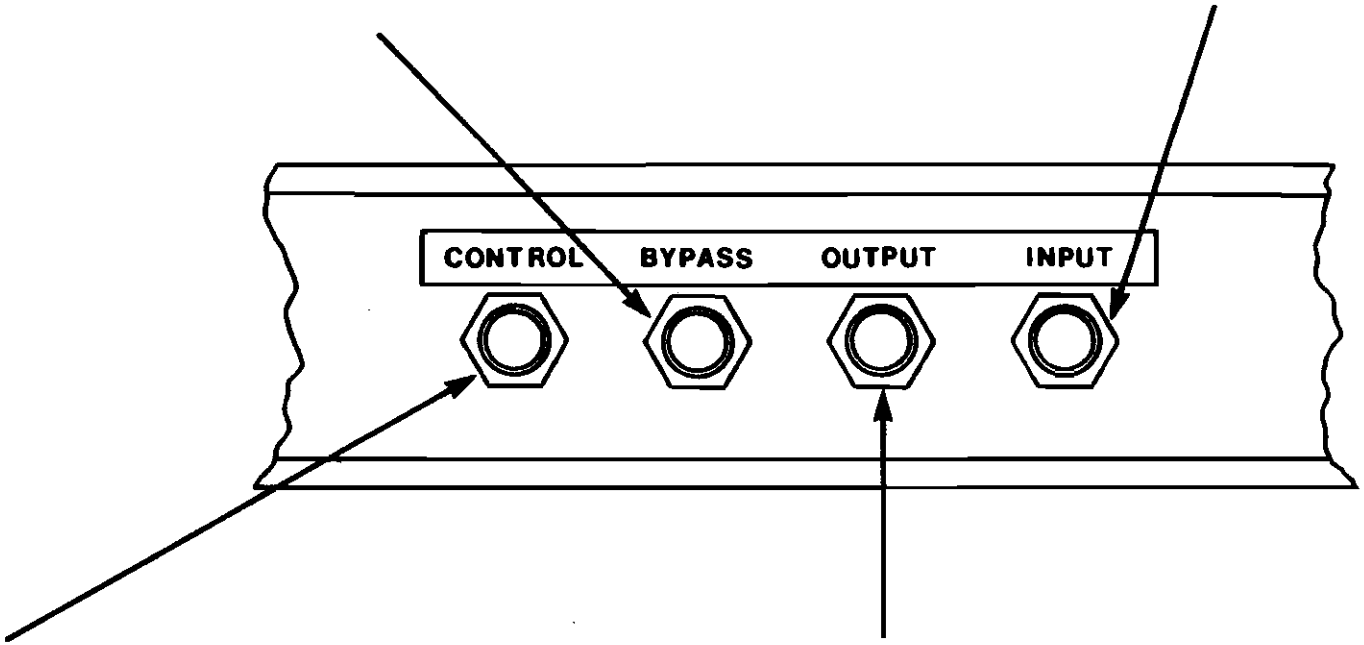
At +100%, the output consists of only the delayed signal (in-phase) Of course, intermediate settings are also possible

BYPASS

The bypass feature of the ADM 64 requires the use of a three conductor (stereo) cable. By shorting the TIP to RING, the signal flows through the cable bypassing the processing circuitry in the unit. The signal, however, still flows through the input and output circuits of the unit.

INPUT

Only one input is provided on the ADM 64. This input accepts signals as low as 0.1Vrms and as high as 8.0Vrms for full dynamic range. The LEVEL control on the front panel is used to set the proper level.



CONTROL

The control jack is used to modulate the internal VCO. A signal generator, envelope follower, synthesizer or footpedal can be used for this purpose. For a two octave sweep, the voltage range is the standard 0 to +5 volts. For a three octave range, the voltage range is extended to -0.62 to +6.2 volts.

NOTE: When the control jack is used, the delay control on the front panel should be in the ENV position. Also, the width and speed controls will interact with the external input. To prevent this interaction, simply set the width control to its MIN setting. This allows the external control to function independently.

OUTPUT

The output consists of both processed and unprocessed signal depending on the setting of the DELAY MIX control on the front panel. At full dynamic range the signal level will be 2.0Vrms nominal. Also, a synthesized stereo output is present on the "ring" portion of the output jack.

INTRODUCTION

The EFFECTRON tm II, ADM 64, is a studio quality, special effects processor designed for the performing musician and is a direct derivative of the ADM 256 and 1024. The ADM 64 uses the latest Adaptive Delta Modulation (ADM) digital encoding technology patented by DELTALAB. Because of DELTALAB's experience and technology you can be assured that your EFFECTRON II will become one of the most important components in your total sound system.

CIRCUIT DESCRIPTION

Even if you are not technically inclined, we urge you to read this chapter. A basic understanding of how signals are processed through the EFFECTRON II will make it easier to predict the effect of control settings and easier to plan control settings to achieve desired sound effects. But, first a word about the "guts" of the design.

Only the latest technology has been used in the design of the EFFECTRON II. This applies to components as well as circuit design.

For example:

1. The power supply uses a toroidal transformer to minimize hum and eliminate stray magnetic fields.

2. The digital memory consists of 16K Dynamic RAM's (Random Access Memory).

3. All digital integrated circuits are CMOS for low power consumption.

4. All analog integrated circuits consist of both bipolar and MOS technology for both low noise and reliability.

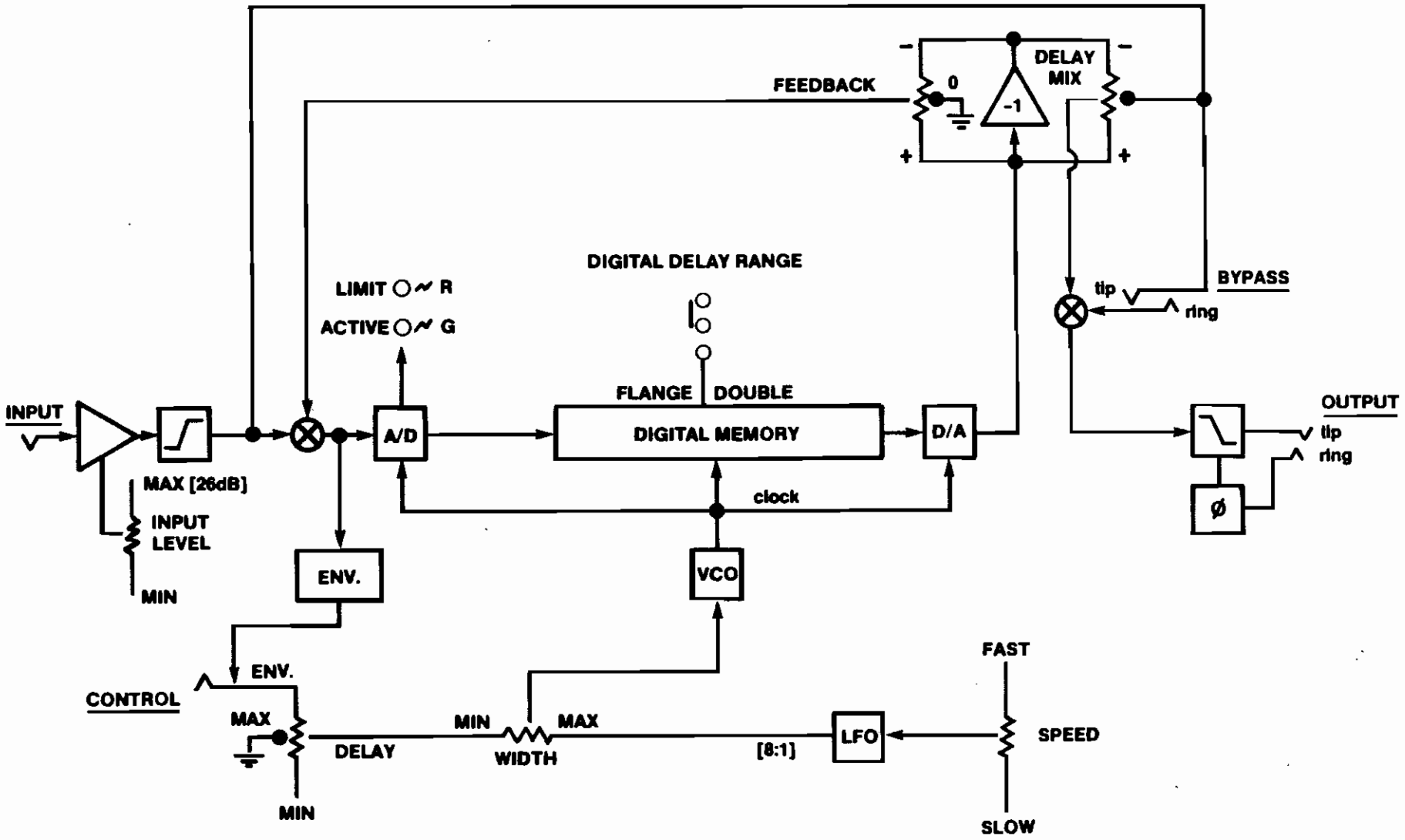
5. Precision resistors (+/-1%) are used in critical circuits to insure repeatability.

6. The printed circuit board is standard mil spec G-10 material.

7. All other components such as potentiometers, switches, etc., have been similarly selected for optimum quality and reliability.

8. For roadability, the chassis is all steel with an aluminum front panel.

The circuit design incorporates DELTALAB's patented Adaptive Delta Modulation technology regarded by audio professionals as the most natural; and cleanest sounding digital encoding technique. A detailed explanation of the circuit design, via the block diagram shown, follows:



EFFECTRON BLOCK DIAGRAM (ADM-64)

The input audio signal is fed into the EFFECTRON II via the INPUT phone jack located on the rear panel. This signal is routed through an input amplifier stage to properly set the operating signal level via the INPUT LEVEL control located on the front panel. The INPUT LEVEL control also sets the gain of the total system (there is no compensating output level control). The minimum input voltage is 0.1 Vrms.

The correct operating level is set by observing the green ACTIVE monitor and the red LIMIT monitor located on the front panel. These LED's monitor the level as seen by the encoding circuits. For maximum dynamic range, the ACTIVE LED should be full-on with a rare flashing of the LIMIT LED during peak passages in the audio input.

Once the proper input level has been set, the audio signal is directed to a pre-emphasis circuit to pre-condition the signal before the actual Analog-to-Digital conversion takes place.

After pre-emphasis the pre-conditioned signal is now sent to the A/D encoder to be converted to a digital signal. It is here that DELTALAB holds all the aces. The Analog-to-Digital encoding scheme is unique. A carefully designed Adaptive Delta Modulator converts the audio signal by analyzing both the value and slew rate of the signal. The result is the most accurate digital representative of the audio signal possible over the full dynamic range.

Next, the digital signal representing the audio is stored in the digital memory. Access to this memory is via the pushbutton switch located on the front panel. Upon selection of a given delay, as indicated by the pushbutton, the digital signal is extracted from the memory and sent to the Digital-to-Analog decoder. Note that the A/D, the memory and the D/A are synchronized to a common clock to insure the proper addressing of the memory banks and to provide the sampling rate for the A/D encoder and D/A decoder.

The D/A decoder is the perfect compliment of the Adaptive Delta Modulation encoder. At the D/A output, the signal is sent to an inverting amplifier and further directed to both the FEEDBACK and DELAY MIX controls located on the front panel.

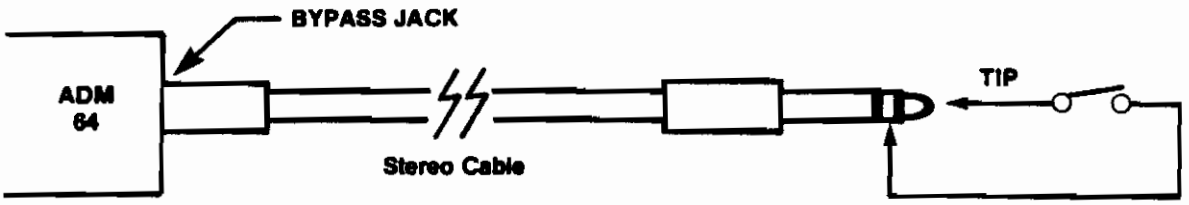
The DELAY MIX control is a center tapped potentiometer, i.e., the unprocessed input signal is applied at the center tap and each phase of the delayed signal is applied at the extreme terminals. As such, a single control is all that is necessary to mix the processed signal (either phase) with the input (source) signal.

The final mix is then de-emphasized to restore the audio to its proper levels. The output is available at the OUTPUT phone jack on the rear panel.

Meanwhile back at the FEEDBACK control, the processed audio (delayed signal) is fed back to a summing point and mixed with the incoming signals at the A/D converter input. The FEEDBACK control, like the DELAY MIX control, is a center tapped potentiometer. The center tap is grounded so that there will be no feedback when the control is in the center position. Again, like the DELAY MIX control, both phases of the processed audio are available at each extreme terminal. Therefore, only one FEEDBACK control is used for

both positive and negative feedback.

The last possible signal path is the BYPASS which requires the use of a stereo (three conductor cable) phone plug to be connected to the BYPASS jack located on the rear panel. By shorting the TIP to the RING, the pre-conditioned audio is directly routed through the cable and back to the de-emphasizing circuitry in the unit to bypass all digital and feedback signal paths. A simple scheme for doing this is shown below:



Switch Closed = Bypass

Switch Open = Normal

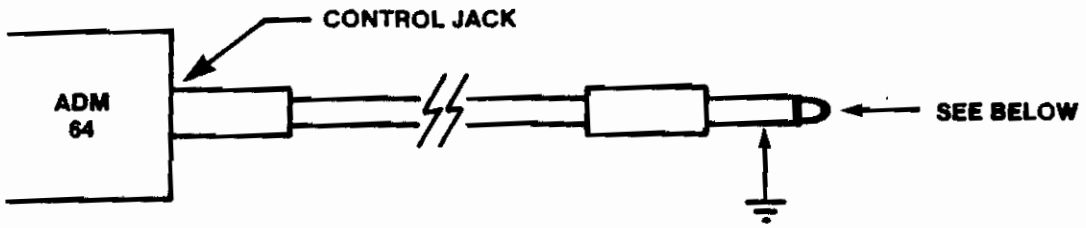
The following functions are not in the signal path but are used to control the basic bit rate clock to create special effects other than those that result by simple feedback. There are three front panel controls whose function is to provide a control voltage into the precision Voltage Controlled Oscillator (VCO) used as the digital bit rate and sampling clock. The VCO requires a voltage of -0.62v to +6.2v to vary the clock in an eight-to-one (three octave) range; and 0v to 5v to vary it four-to-one (two octaves).

Setting the WIDTH control, located on the front panel, in the MIN position, the DELAY control, also located on the front panel, provides the required voltages as shown in the block diagram for the first half of its range. Varying the delay from the Center position clockwise activates the Envelope Follower control signal. For maximum Envelope Follower action, the WIDTH control should be set to MIN; however, a variety of effects can be derived by allowing the WIDTH control to mix both the Envelope Follower and Low Frequency Oscillator (LFO) signals. It should be noted that there is an interaction between all three of the above mentioned controls. For example, when the delay control is in an intermediate setting, the internal LFO is mixed with the delay control to produce a more natural doubling effect.

When an external signal is applied to the CONTROL jack located on the rear panel, the Envelope Follower is disabled. The DELAY control should be full clockwise for maximum external control. A basic scheme for supplying an external control voltage is shown on the next page.

The WIDTH control is used in conjunction with the SPEED control located on the front panel to apply a Low Frequency Oscillating (LFO) voltage to the VCO input. The LFO is a sinusoid whose frequency is controlled by the SPEED control. The WIDTH control determines the amplitude fed into the VCO. It

can be seen that intermediate settings of the WIDTH control will cause a mix of both the DELAY setting and LFO output. Also, when a voltage is applied to the external CONTROL jack, the WIDTH will likewise mix internal LFO settings with the external control voltage — all of which makes for interesting effects.



RANGE	EXTERNAL INPUT
4;1	0 V to 5 V
8;1	-0.62 V to 6.2 V

The EFFECTRON II is basically a simple to understand device, but because of its simplicity it is a very power effects tool.

INITIAL SET UP

Due to its simplicity there are no special setup rules that need to be observed. We do; however, recommend the following be used as a starting point until you familiarize yourself with each control:

INPUT LEVEL	-	MIN: Full counter clockwise
FEEDBACK	-	0: Center position
DELAY (BUTTON)	-	Start with button "in"
DELAY	-	MAX position: Center detent
WIDTH	-	MIN: Full counter clockwise
SPEED	-	Slow: Full counter clockwise
DELAY MIX	-	Source: Center position

The INPUT LEVEL can now be adjusted for proper level via the LED monitors. All other functions can be varied at will to create the various effects.

NOTE: IN ORDER TO INSURE MAXIMUM RELIABILITY AND PREVENT PREMATURE FAILURE DUE TO THERMAL SHOCK I.E., HOT AND COLD VARIATIONS THE ADM64 DOES NOT HAVE AN OFF/ON SWITCH. THE AVERAGE POWER CONSUMPTION IS LESS THAN 5 WATTS AND SHOULD NOT AFFECT THE OPERATING ECONOMY OF YOUR TOTAL SOUND SUSTEM.

APPLICATIONS

While the EFFECTRON II is capable of providing a very large array of effects, they all fall into the basic categories (or combination of these) outlined below:

1. STRAIGHT DELAY

    a. DOUBLING,, THICKENING, BROADENING
    b. PRE-REVERB DELAY
    c. HAAS-EFFECT IMAGE SHIFTING

2. FEEDBACK OF SHORT DELAYS

    a. COMB FILTERING
    b. FLANGING
    c. FLANGING WITH FEEDBACK
    d. FEEDBACK PHASE INVERSION
    e. TUNED RESONANCE

3. TIME BASE MODULATION

    a. MANUAL PITCH SHIFTING
    b. VIBRATO, AUTOMATIC PITCH SWEEPING
    c. ENVELOPE FOLLOWER

1. STRAIGHT DELAY

Sound travels in air at a finite speed — approximately 340 meters per second (1100 feet per second), or about one foot per millisecond — and many of the uses of a digital delay processor involve the controlled electronic recreation of effects which occur in acoustics due to this finite speed.

For instance, for practical reasons, vocals and instrumentals are usually recorded in a nearly anechoic fashion by close-miking in an acoustically absorptive studio, but this sometimes yields an anemic, uninteresting sound. Whenever we hear vocal instrumental sounds in a living room or concert environment, the dry sound is accompanied by reflections off nearby walls, floor and furniture. These early reflections accompanying the direct sound — slightly delayed because of their longer airpath — add apparent volume and fullness and thus enrich the character of the sound.

The subjective effect of reflections (delayed replicas of an original sound) depends on the length of the delay as follows:

— Single or multiple delays up to about 40 milliseconds after the direct sound alters the apparent character of quality, but are not perceived separately. Typically, they add the sort of "fullness" and body which a

voice has in a living room, but lacks when heard outdoors. They are called "early" reflections.

— A single delay within about 40 milliseconds and having the same volume level as the direct sound, produces an effect something like that heard when a solo voice is replaced by two identical voices singing in unison. This is called "doubling."

— A single delay longer than 40 to 50 milliseconds starts to break away from the original sound and may be perceived as an echo.

a. DOUBLING, THICKENING, BROADENING

Two voices singing together, or a single voice which is double-tracked (overdubbed to accompany its previously recorded self) produce a combined sound which is richer and more interesting than simply turning up a single vocal track 3 db in level. One reason is that the two separate voices are never recorded in exactly precise synchronism. The waveform of one is usually a few milliseconds ahead or behind the other despite the most careful of rehearsals.

This effect can be simulated with any signal source simply by delaying it by 16 to 64 milliseconds and mixing the delayed signal with the original at equal levels. This is called doubling. It thickens the texture producing a more "full-bodied" sound while it increases the apparent loudness without significantly raising VU meter levels. It is particularly useful for adding strength and character to a thin-sounding vocal.

You should experiment with the length of the delays used for doubling. With delays in the 16-20 mS range, the sound remains relatively tight and focused. Delays of 30-40 mS produce a more obvious broadening. Doubling with delays shorter than about 16 mS is not recommended because of the risk of coloration due to comb filtering. If you extend the delay beyond about 40 mS it may be heard as a distinct echo.

This process simulates two vocal tracks with a small, but constant delay between them. Of course, when real voices are recorded or overdubbed, they have varying small differences between them. So, to make its doubling seem more realistic, use the MODULATOR to continuously vary the delay. The action must be subtle so as to avoid audible pitch wobble. Setting the width and speed controls both to between 7:00 and 9:00 o'clock, usually produces a pleasingly realistic doubling effect with no audible side effects. Also, the Envelope Follower, selected by the DELAY control can be similarly used to produce detuned transient realism.

b. PRE-REVERB DELAY

Authentic acoustic reverberation in a large space requires many dozens of milliseconds to buildup. But, in many studio reverb units (spring, plate or acoustic chamber) an output signal begins

to appear very rapidly after the onset of the input signal. The subjective performance of reverberators is usually improved by delaying the signal fed to them. The delays used for doubling often serve as pre-reverb delays.

In a concert environment, the listener hears the direct sound from the performer, followed by "early" reflections from the floor and walls of the stage and finally the reverberation comprised of long delays as sound is reflected among the walls of the auditorium. When using a reverb system to process dry studio sound, pre-reverb delay can also serve as the "early" stage-area reflections, simply by including them in the mix.

c. HAAS-EFFECT IMAGE SHIFTING

The ear has an integration time of about 40 mS. If a sound is heard from one direction and the same sound also arrives at a later time from another direction, the second arrival will not be heard at all if the interval between the two arrivals is less than 40 mS. The later arriving sound may even be several dB higher in level than its earlier counterpart; nevertheless, the ear will hear both sounds as a single louder sound located in the direction of the first arrival. This is the precedence effect — or Haas effect — and it can be used to stabilize images in a stereo sound field.

For example, if a signal is recorded in both channels at identical levels, but the feed to the right channel is delayed, then the sound will be perceived as originating exclusively in the left channel. In principle, the delay to the opposite channel may be anything from 1 mS to 40 mS. But, long delays carry the risk that some listeners may begin to perceive the delayed sounds as an echo; and if the delay is under 10-15 mS, coloration due to comb filtering could become a problem if the two channels are later mixed together (for AM broadcast, for instance). A delay of around 20 mS usually turns out to be optimum for Haas-effect image panning.

It is possible to keep the signal level constant in both channels and cause the image to jump back and forth from left to right by swapping the delayed and undelayed signals by panning the mixer.

2. FEEDBACK OF SHORT DELAYS

By using the feedback control, the delayed signal is fed back and mixed with the incoming audio signal and then the composite signal is encoded and read into the digital delay circuits. The strength and tonal quality of the feedback may be adjusted by the user. The maximum feedback gain is less than unity; thus, like a read sound reflecting off any surface, the recycled signal is at least a little weaker than the original and as the sound is repetitively recycled through the system, it gradually fades away.

a. COMB FILTERING

Whenever any original sound and a delayed version of itself are mixed together, the two sounds are mutually reinforced at some frequencies and tend to mutually cancel at other frequencies. This occurs in an electronic delay line and it also is a common acoustic phenomenon both in recording (where the mike picks up both the direct sound and a reflected sound from the floor, wall or music stand) and in playback (where a loudspeaker's direct sound combines in the air with reflections off room boundaries and furnishings). This pattern of alternating reinforcement and cancellation causes an audible coloration of the sound.

Two things are noteworthy. One is that the reinforcements and cancellations occur at harmonically-related frequencies. The other is that the pattern of peaks and dips can be varied, i.e., tuned, just by varying the delay time. In almost all normal vocal and musical sounds, most of the energy is found at fundamental frequencies and their harmonic overtones. By varying the delay time of a delay-and-mix circuit, we can easily but dramatically alter the overtone structure and the tonal quality of any steady sound. The precise delay times and unusually flat frequency response of the EFFECTRON II make it ideal for comb filtering on demand.

b. FLANGING

If the unit is setup to produce comb filtering and then the delay time is smoothly varied, the pattern of peaks and nulls will shift in frequency. Modulate the delay time rapidly and the pattern of peaks and nulls will sweep rapidly up and down the frequency spectrum passing in and out of synchronization with the frequencies of musical signals and their overtones. Select relatively short delays so that the spacing of the nulls corresponds to that of musical harmonics. As the delay is swept, at one moment the odd-numbered harmonics are enhanced by 6 dB — a moment later, the harmonic structure of that note is sliding into alignment with the harmonic overtones of a different note. This sweeping, shifting change is called "flanging."

To achieve it, one need only setup the delay and mixing as required for comb filtering and then activate the modulator to sweep the delay time up and down. Typically the modulator WIDTH is set at maximum in order to produce a broad sweep and the modulator SPEED is set at a modest value (e.g., between 7:00 and 9:00 o'clock) so that the comb filter sweeps up and down every few seconds. A high setting of the modulator SPEED would sweep the notch pattern over the musical spectrum too rapidly for the changing harmonic structure of the music to be heard and might also cause audible pitch wobble of the musical signal itself. You should experiment to find the settings of DELAY, WIDTH, SPEED and mixing ratio which yield interesting flanging effects with various vocal and instrumental sounds. In general, higher-pitched sounds work best with shorter delays.

c. FLANGING WITH FEEDBACK

Mixing the dry and delayed signals at approximately the same level causes comb filtering (a pattern of nulls and peaks in frequency response). Modulation of the delay time causes the pattern of nulls and peaks to sweep through the musical spectrum producing "flanging."

A stronger, richer flange is reproduced by using maximum feedback to create a chain of closely-spaced delays whose uniform spacing sharpens and strengthens the comb filter peak and nulls. Select short delays such as 1 mS or 4 mS and set the FEEDBACK near the maximum setting, either positive or negative. Finally, modulate the delay by setting modulator WIDTH at maximum and SPEED between 7:00 o'clock (for a slow flange) and 12:00 o'clock (for a fast flange). Note that it is important to use near maximum feedback so that each successive recycled delay is at nearly the same strength as its predecessor, yielding the strongest reinforcement/cancellation pattern. The flange can be varied, of course, by selecting different front panel delays. If the WIDTH control is set less than maximum, then the DELAY can be used to alter the delays and thus shift the frequencies of the comb-filter pattern. Finally, the flanged signal must be mixed with the dry signal via the delay mix control.

d. FEEDBACK PHASE INVERSION

The feedback phase control gives you the option of mixing the feedback delays either in phase (+) or out of phase (-) with the incoming dry signals. This choice alters the frequency distribution of any comb-filtering coloration which may occur as a result of combining dry and delayed signals. You can make this choice by ear. When using feedback, simply rotate the feedback control to find out whether positive or negative feedback sounds better in each particular situation.

e. TUNED RESONANCE

Push delay button out setting delay control approximately half-way to produce comb-filtering and then use maximum feedback to create a string of closely spaced delays whose uniform spacing sharpens and strengthens the peaks and nulls of the comb-filter pattern. As a result, some of the peaks will be fed back at almost sufficient strength to cause a sustained feedback oscillation. Whenever the system is stimulated by a broadband input signal such as drums or a speaking voice, it will tend to "ring" at frequencies of the comb-filter peak. Since the peaks in a comb-filter form a harmonic series, a distinctly musical tonality will be produced. By varying the delays, the pitch of the tuned resonance can be tuned as desired. Inverting the polarity of the feedback will also alter the pitch of the resonance. For best results, the FEEDBACK control will normally have to be at a maximum setting.

A principal application of tuned resonance is the processing of human speech to produce "computer speech" by adding a fixed-pitch metallic resonance. Such resonances can also be used to color various percussive sounds in interesting ways.

The pitch of the tuned resonance may be varied dynamically by means of the VCO — typically by setting WIDTH close to maximum SPEED between 9:00 and 12:00 o'clock. An unpitched broadband sound such as a repeating drumbeat can be made to sound rather like a guitar.

To create a string of echos, setup a long delay (over 40 mS) to produce a discrete echo and then use feedback to recycle the signal repetitively through the delay. In general, echos are more interesting if they are separated spatially from the dry source and from each other by panning at the mixing console.

3. TIME BASE MODULATION

Much of the EFFECTRON II's flexibility and usefulness as a studio or onstage tool arises from its ability to vary the speed of the "clock" which governs how rapidly signals are shuttled through the digital memory, under either manual or automatic control.

The effects obtainable by varying the clock are easily understood by analogy with a tape recorder having variable tape speed whose recording and playback heads are separated by some distance. The time-delay between recording and playback is governed by the separation of the heads and by the tape speed. If the speed is doubled, the tape will traverse the distance in half the time. As long as the tape speed remains constant while the tape is being recorded and played, then the tape speed will affect only the delay. But, if the tape is recorded at 7 1/2 ips and then the speed is doubled before the tape arrives at the playback head, the waveforms will pass the playback head twice as rapidly as they passed the recording head and the frequency of each sound will be doubled, i.e., the musical pitch will rise an octave. Thus a change in tape speed which occurs in the interval between recording and playback alters not only the delay time, but also the pitch — delay modulation causes pitch modulation.

a. MANUAL PITCH SHIFTING

The DELAY adjust control adjusts the clock speed over a 4-to-1 range and consequently is capable of varying the pitch of a musical tone over a 4-to-1 range in frequency — or two full octaves in pitch.

Any signal passing through will change in pitch as the DELAY is changed. As noted earlier, the pitch shift depends on the change in clock speed. The faster the DELAY knob is turned, the greater the pitch shift will be and the longer the delay time is, the easier it will be to alter the clock speed significantly during the delay. The pitch is shifted downward as the DELAY is increased and vice versa.

Gradual pitch changes are sometimes referred to as Doppler shifts, an analogy to the Doppler effect which occurs with moving sound sources such as train and auto horns (the pitch rises when a sound source is moving toward the listener and falls as the source recedes).

b. VIBRATO, AUTOMATIC PITCH SWEEPING

It is difficult to accomplish really smooth pitch changes by manually rotating the DELAY adjust control so the modulator function provides an automatic method of achieving smooth pitch modulation effects. The LFO modulates the internal clock up and down by an amount set by the WIDTH control and at a rate set by the SPEED control.

As with manual pitch shifting, the amount of modulator-actuated frequency shift will depend on the choice of delay time as well as on the modulator controls. With delay of a few milliseconds and with the WIDTH and SPEED controls set at about 10:00 o'clock, a subtle pitch modulation is produced which most listeners can just detect. Lengthening the delay or increasing the WIDTH yields a larger and more obvious sweeping of the pitch up and down. Increasing the SPEED beyond 12:00 o'clock, not only sweeps the pitch up and down more rapidly, but also sweeps it over a wider frequency range — up to several octaves.

Vibrato is a rapid, low-amplitude pitch modulation which is attractive with most vocals and with some instrumental sounds. It must be used with caution lest it sounds ludicrous — excessively wide vibrato will simply be perceived as an off-key pitch wobble. Typical control settings for producing an authentic-sounding and attractive vibrato are: DELAY 32 mS, WIDTH between 7:00 and 9:00 o'clock, SPEED between 1:00 and 3:00 o'clock.

c. ENVELOPE FOLLOWER

By detecting the amplitude of the input signal, the envelope follower is used to modulate the clock. This results in a variable time delay that varies with the amplitude of the signal. Using this type of clock modulation produces flanging effects that "sweep up" with signal decay. The effect is most pleasing when used with drum and guitar tracks.

Also, in the doubling mode, the envelope follower can "detune" the doubling effect during transients to create a very natural effect. Remember, two vocalists do not always sing in perfect tune. The ADM 64 can simulate a real situation.

SPECIFICATIONS

DELAY RANGE	MIN	TO	MAX
Flange	1mS	to	4mS
Double	16mS	to	64mS
FREQUENCY RESPONSE			
+1, -3dB @ -10dB below LIMIT	20 to 16kHz @ all delay settings		
DYNAMIC RANGE			
A - weighted	95dB typ., 90dB min		
DISTORTION			
Ref 1kHz @ LIMIT	0.2% max		
INPUT RANGE	-26 to +17dBV (0.05 to 7.1VrmS)		
INPUT LEVEL GAIN			
Min	---- (X0.0 gain)		
Unity @ (Center Position)	0dB (X1.0 gain)		
Max	+26dB (X20 gain)		
OUTPUT LEVEL (@ LIMIT)			
METERING			
LIMIT	0dB ref		
ACTIVE	-20dB below ref		
MODULATION			
WIDTH (8-to-1)	MIN	TO	MAX
Flange	0.9mS	to	7.2mS
Double	14.5mS	to	116mS
SPEED	0.05Hz	to	10Hz nominal
CONTROL (External)			
(4-to-1 range)	0	to	5.0V
(8-to-1 range)	-0.62V	to	6.2V
ENVELOPE FOLLOWER	Modulates with respect to amplitude		
FEEDBACK	Recirculates delayed signal		
BYPASS (EXTERNAL)	Shorted - bypasses total system; Open - normal operation		
SIZE	1 3/4x19x7in (4.45x48.3x17.8cm)		
SHIPPING WEIGHT	10 lbs		

Manufacturer reserves the right to make improvements without notice or obligation; therefore, all specifications are subject to change.

IN CASE OF DIFFICULTY

In many instances where difficulty is experienced, it is best to check out a few simple questions before "panic" sets in:

1. Is the unit plugged into the power line? The power (Yellow LED) monitor on the front panel should indicate this.

2. Are all inputs, outputs and control signals connected to their respective jacks on the rear panel?

3. If you are using an external bypass switch, have you placed the unit in the bypass mode? This would totally prevent any kind of signal processing.

4. Are the input signal levels in the proper range? The INPUT LEVEL control and the ACTIVE/LIMIT LED's will properly set the input for optimum dynamic range. (Low noise, low distortion, etc.)

5. If this is the first use of the EFFECTRON II, have you referred to the initial setup as suggested in the Owner's Manual?

Every effort has been made to insure trouble-free performance from each EFFECTRON II. Should a problem occur, simply call collect and ask for Bruce Wayne at (617) 256-9034. Should this be impractical, notify us by writing to:

DELTALAB RESEARCH, INC.
ATTN: Customer Service
19 Alpha Road
Chelmsford, MA 01824

Describe the nature of the problem, the steps you have taken to diagnose it, the serial number of the unit and whether or not you have retained the original shipping carton.

NOTE: If the unit must be returned to our factory, we will provide you with a Return Authorization (RA) Number which must be prominently displayed on the outside of the shipping carton. Any unit which arrives without a visible RA Number may be refused by our Receiving Department.

LIMITED WARRANTY

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DELTALAB RESEARCH, INC., ("DELTALAB") warrants to the first purchaser of a new DELTALAB EFFECTRON II, that the unit is free from defects in material and workmanship. DELTALAB's sole obligation under this warranty shall be to provide, without charge, parts and labor necessary to remedy defects, if any, which appear within ninety (90) days from the date of purchase.

This warranty is the sole and exclusive express warranty given with respect to the unit and all other express warranties are hereby excluded. Implied warranties, including those of merchantability and fitness for a particular purpose, are limited to ninety (90) days from the date of purchase. SOME STATES DO NOT ALLOW LIMITATIONS ON HOW LONG AN IMPLIED WARRANTY LASTS, SO THE ABOVE LIMITATIONS MAY NOT APPLY TO YOU. DELTALAB IS NOT RESPONSIBLE FOR INDIRECT, INCIDENTAL OR CONSEQUENTIAL DAMAGES. SOME STATES DO NOT ALLOW THE EXCLUSION OR LIMITATION OF INCIDENTAL OR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATION OR EXCLUSION MAY NOT APPLY TO YOU.

This warranty does not apply if the unit has been:

(1) Repaired, worked on or altered by persons unauthorized by DELTALAB in such a manner as to injure, in DELTALAB's sole judgment, the performance, stability or reliability of the unit;

(2) Subjected to misuse, negligence or accident; or

(3) Connected, installed, adjusted or used otherwise than in accordance with the instructions furnished by DELTALAB.

This warranty is valid only when the unit is returned to DELTALAB within ninety (90) days from the date of purchase, two-way freight prepaid, together with a copy of the original invoice from the authorized DELTALAB dealer.

This warranty gives you specific legal rights, and you may also have other rights which vary from state to state.

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TYPICAL SET UPS

EFFECT: COMB FILTERING

Setting the RANGE button to the "FLANGE" (out) position results in a comb filter frequency response. FEEDBACK increases intensity. Phase reversal in MIX or FEEDBACK (if used) changes character of sound.

EFFECT: FLANGING

Set RANGE button to the "FLANGE" (out) position. Experiment with WIDTH and SPEED to taste. The MAX width ratio is 8:1. Note the differences in MIX being in negative versus positive settings.

EFFECT: FLANGING WITH FEEDBACK

Set the RANGE button to the "FLANGE" (out) position. Experiment with FEEDBACK, WIDTH, SPEED AND MIX to taste. Note the differences in negative and positive feedback and mix and combinations thereof.

EFFECT: ENVELOPE FLANGE

Set the RANGE button to the "FLANGE" (out) position. Experiment with the DELAY control from its MAX setting on clockwise to the full ENV (envelope) setting. Note the difference in negative and positive MIX.

EFFECT: ENVELOPE FLANGE WITH FEEDBACK

Set the RANGE button to the "FLANGE" (out) position. Use the DELAY control from the MAX thru ENV setting and add FEEDBACK to taste. Note differences in negative and positive feedback and mix.

EFFECT: ENVELOPE / SWEEP FLANGE COMBINATION

Set the RANGE button to the "FLANGE" (out) position. Use the DELAY control in its MAX-ENV range and blend in the WIDTH control to taste. Note differences in negative and positive mix.

EFFECT: ENVELOPE / SWEEP FLANGE COMBO WITH FEEDBACK

Set the RANGE button to the "FLANGE" (out) position. Use the DELAY control in its MAX-ENV range and blend in the WIDTH, SPEED and FEEDBACK controls to taste. Note differences in negative and positive settings of feedback and mix.

EFFECT: FLANGE VIA EXTERNAL CONTROL (REAR PANEL)

Set the RANGE button to the "FLANGE" (out) position. Set DELAY control full clockwise to the ENV position. Set WIDTH and SPEED full counter clockwise. Note differences in negative and positive settings of feedback and mix. Also, note that DELAY, WIDTH and SPEED can be varied to combine internal controls with the external control input.

EFFECT: BASIC DOUBLING

Set the RANGE button to the "DOUBLE" (in) position. Vary the DELAY control from its MIN to MAX setting to taste. Note that intermediate settings of the DELAY control automatically adds a controlled amount of modulation for realism, even though the WIDTH control is at MIN. Vary SPEED and FEEDBACK to taste for multi-track effects.

EFFECT: DETUNED DOUBLING

Set the RANGE button to the "DOUBLE" (in) position. Use the DELAY control in its MAX-ENV range to vary the delay with respect to the amplitude of the signal. This will detune the doubling effect during transients much like "real life."

EFFECT: ASYNCHRONOUS DOUBLING

Set the RANGE button to the "DOUBLE" (in) position. By mixing the modulator cotrols, SPEED and WIDTH, with the envelope follower (DELAY control in MAX-ENV range), the result is a totally natural, non-synchronous doubling that detunes with transients and remains unsynchronized. Careful experimentation will yield amazingly natural results; be careful not to overdo it.

EFFECT: PRE-REVERB DELAY

Set the RANGE button to the "DOUBLE" (in) position. A pre-reverb delay of 16 mS to 64 mS is possible by varying the DELAY control from its MIN to MAX range. Note that intermediate settings will include some LFO modulation — as much as 10%. By keeping the speed control fully counter clockwise, the modulation mix should not be objectionable.

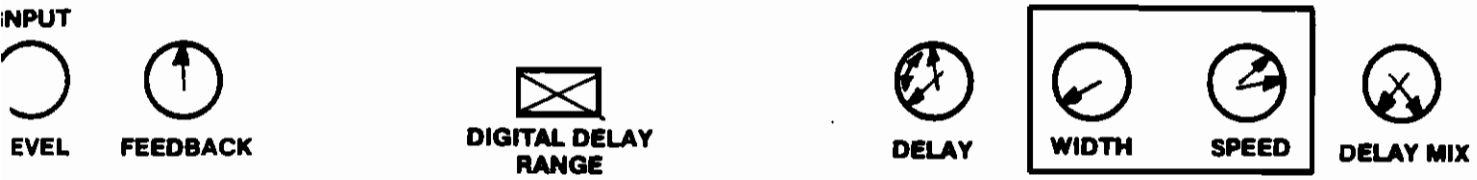
EFFECT: TUNED RESONANCE

Two ranges are possible:

RANGE button in "FLANGE" (out) position
1 mS to 4 mS.

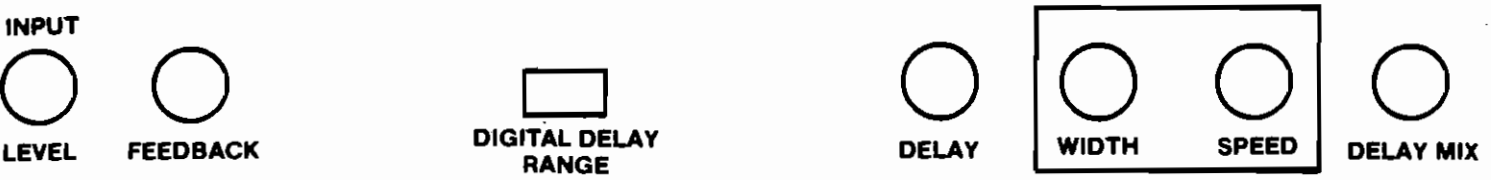
RANGE button in "DOUBLE" (in) position
16 mS to 64 mS.

Use maximum FEEDBACK either positive or negative. Set MIX to either +100% or -100%. Vary the DELAY control to set reverberant pitch desired.



EFFECT: VIBRATO

Set the RANGE button to the "DOUBLE" (in) position. Experiment with the SPEED, WIDTH, DELAY and FEEDBACK controls to taste. Note that the process also adds a little tremolo with the resulting vibrato.



EFFECT: _____

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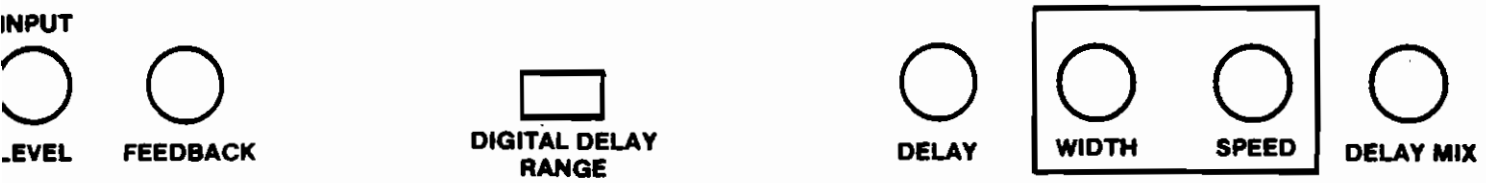

EFFECT: _____

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